Features and Software options

Package Name	Number	Mnemonic	Release
1.5 Mbit Digital Trunk Interface	75	PBXI	5
— Hong Kong Digital Trunk Interface			
— Reference Clock Switching (See also packages 129, 131, and 154)			
16-Button Digitone/Multifrequency Telephone	144	ABCD	14
— 16-Button Digitone/Multifrequency Operation			

Package Name	Number	Mnemonic	Release
2 Mbit Digital Trunk Interface	129	DTI2	10
• DID Recall features on DTI2 for Italy – DID Offering			
— DID Recall features on DTI2 for Italy – DID Recall			
— Italian Central Office Special Services (see also packages 131, and 157)			
— Italian Periodic Pulse Metering			
— Pulsed E&M DTI2 Signaling			
— Reference Clock Switching (see also packages 75, 131, and 154)			
— R2MFC 1.5 Mbps DTI			
— 2 Mbps Digital Trunk Interface			
— 2 Mbps Digital Trunk Interface Enhancements:			
• Alarm Handling on DID Channels			
• Alarm Handling on Incoming COT/DID Calls			
• Call Clearance			
• Clock Synchronization			
• DID Call Offering			
• Disable Out-of-Service Alarm State			
• Fault Signal			
• Incoming Seizure			
• Outpulsing Delay			
• Release Control			
• Signal Recognition			
• Trunk Entering Alarm Status/Trunk Pack Exiting Alarm Status			
• 64 Kbps Alarm Indication Signal (AIS) Handling			
2.0 Mb/s Primary Rate Interface	154	PRI2	14
— Reference Clock Switching (see also packages 75, 129, and 131)			

Package Name	Number	Mnemonic	Release
2500 Set Features	18	SS25	1
— Call Hold, Permanent			
— 2500 Set Features			
500 Set Dial Access to Features	73	SS5	4
— 500 Set Features			
— 500/2500 Line Disconnect			
AC15 Recall	236	ACRL	20
— AC15 Recall: Timed Reminder Recall			
— AC15 Recall: Transfer from Norstar			
— AC15 Recall: Transfer from Meridian 1			
— Access Restrictions			
ACD/CDN Expansion	388	ACDE	25.40
— ACD/CDN Expansion			
Administration Set	256	ADMINSET	21
— Set-based Administration Enhancements			
Advanced ISDN Network Services	148	NTWK	13
— Advice of Charge – Charging Information and End of Call for NUMERIS Connectivity (see also package 101)			
— Advice of Charge Real-time Supplementary Services for NUMERIS and SWISSNET (see also package 101)			
— Alternative Conference PAD Levels			
— Alternative Loss Plan			
— Alternative Loss Plan for China			
Analog Calling Line Identification	349	ACLI	25
— CLID on Analog Trunks for Hong Kong (A-CLID)			

Package Name	Number	Mnemonic	Release
Aries Digital Sets	170	ARIE	14
— Meridian Communications Adapter			
— Meridian Modular Telephones			
Attendant Administration	54	AA	1
— Attendant Administration			
Attendant Alternative Answering	174	AAA	15
— Attendant Alternative Answering			
— Attendant Barge-In			
Attendant Announcement	384	AANN	25.40
— Attendant Announcement			
Attendant Break-In/Trunk Offer	127	BKI	1
— Attendant Break-In			
— Break-In busy Indication and Prevention			
— Break-In to Inquiry Calls			
— Break-In to Lockout Set Denied			
— Break-In with Secrecy			
— China Number 1 Signaling – Toll Operator Break-In (see also Package 131)			
— Network Individual Do Not Disturb (see also packages 9, and 159			
— Attendant Busy Verify			
— Attendant Call Selection			
— Attendant Calls Waiting Indication			
— Attendant Consoles			
— Attendant Delay on Hold			
— Attendant Display of Speed Dial or Autodial			

Package Name	Number	Mnemonic	Release
Attendant Forward No Answer	134	AFNA	14
— Attendant Forward No Answer			
— Attendant Forward No Answer Expansion			
— Attendant Incoming Call Indicators			
— Attendant Interpositional Transfer			
— Attendant Lockout			
Attendant Overflow Position	56	AOP	1
— Attendant Overflow Position			
— Attendant Position Busy			
— Attendant Recall			
— Attendant Recall with Splitting			
Attendant Remote Call Forward	253	ARFW	20
— Call Forward, Remote (Network and Attendant Wide)			
— Attendant Secrecy			
— Attendant Splitting			
— Attendant Trunk Group Busy Indication			
— Audible Reminder of Held Calls			
Autodial Tandem Transfer	258	ATX	20
— Autodial Tandem Transfer			
Automated Modem Pooling	78	AMP	5
Automatic Answerback	47	AAB	1
— Automatic Answerback			
— Automatic Call Distribution Answer Time in Night Service			

Package Name	Number	Mnemonic	Release
— Automatic Call Distribution Call Delays (see also package 40)			
— Automatic Call Distribution Call Priority (see also package 40)			
— Automatic Call Distribution Call Waiting Thresholds (see also packages 40 and 41)			
— Automatic Call Distribution Calls on Hold (see also package 40)			
— Automatic Call Distribution Dynamic Queue Threshold (see also package 40)			
Automatic Call Distribution Enhanced Overflow	178	EOVF	15
— Automatic Call Distribution Enhanced Overflow			
Automatic Call Distribution Load Management	43	LMAN	1
— Automatic Call Distribution Load Management Reports			
Automatic Call Distribution Night Call Forward without Disconnect Supervision	289	ADSP	23
— Call Processor Input/Output)			
Automatic Call Distribution Package C	42	ACDC	1
— Automatic Call Distribution Report Control (see also package 50)			
— 500/2500 Line Disconnect			
Automatic Call Distribution Package D, Auxiliary Link Processor	51	LNK	2
— ACD Package D Auxiliary Processor Link			
Automatic Call Distribution Package D, Auxiliary Security	114	AUXS	12
— ACD-D Auxiliary Security			

Package Name	Number	Mnemonic	Release
Automatic Call Distribution Package D	50	ACDD	2
— Automatic Call Distribution Report Control (see also package 42)			
— Automatic Call Distribution Threshold Visual Indication (see also packages 40 and 41)			
Automatic Call Distribution, Account Code	155	ACNT	13
— Automatic Call Distribution Activity Code			
Automatic Call Distribution, Package A	45	ACDA	1
— Automatic Call Distribution			
Automatic Call Distribution, Package B	41	ACDB	1
— Automatic Call Distribution Call Waiting Thresholds (see also packages 40, and 131)			
— Automatic Call Distribution Least Call Queuing			
— Automatic Call Distribution Threshold Visual Indication (see also packages 40, and 131)			
Automatic Call Distribution, Priority Agent	116	PAGT	12
— Automatic Call Distribution Priority Agent			
Automatic Call Distribution, Timed Overflow Queuing	111	TOF	10
— ACD Timed Overflow			
— Automatic Gain Control Inhibit			
— Automatic Guard Detection			
— Automatic Hold			
Automatic ID of Outward Dialing	3	AIOD	1
Automatic Installation (Option 11 only)	200	AINS	16
— Automatic Installation			
Automatic Line Selection	72	LSEL	4
— Automatic Line Selection			

Package Name	Number	Mnemonic	Release
Automatic Number Identification Route Selection	13	ANIR	1
— Automatic Number Identification Route Selection			
Automatic Number Identification	12	ANI	1
— Automatic Number Identification			
— Automatic Number Identification on DTI			
— Automatic Preselection of Prime Directory Number			
Automatic Redial	304	ARDL	22
— Automatic Redial			
— Automatic Timed Reminders			
Automatic Wake-Up	102	AWU	10
— Automatic Wake Up			
Auxiliary Processor Link	109	APL	10
— Auxiliary Processor Link			
— Auxiliary Signaling			
— B34 Dynamic Loss Switching (see also packages 164 and 203)			
Background Terminal	99	BGD	10
— Background Terminal Facility			
Basic Alternate Route Selection	57	BARS	1
— Network Alternate Route Selection/Basic Alternate Route Selection Enhancement – Local Termination (see also package 58)			
Basic Authorization Code	25	BAUT	1
— Basic Authorization Code			

Package Name	Number	Mnemonic	Release
Basic Automatic Call Distribution	40	BACD	1
— Automatic Call Distribution Alternate Call Answer			
— Automatic Call Distribution Call Delays (see also package 131)			
— Automatic Call Distribution Call Priority (see also package 131)			
— Automatic Call Distribution Call Waiting Thresholds (see also packages 41, and 131)			
— Automatic Call Distribution Calls on Hold (see also package 131)			
— Automatic Call Distribution Dynamic Queue Threshold (see also package 131)			
— Automatic Call Distribution Enhancements			
— Automatic Call Distribution in Night Service			
— Automatic Call Distribution Threshold Visual Indication (see also packages 41, and 131)			
— INIT Automatic Call Distribution (ACD) Queue Call Restore			
Basic Call Processing	0	BASIC	1
Basic Queuing	28	BQUE	1
— Basic Queuing			
Basic Rate Interface	216	BRI	18
— Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 235)			
Basic Routing	14	BRTE	1
— Basic Routing			
Boss Secretary Filtering (FFC activation)	198	FTCSF	15
— Flexible Feature Code Boss Secretarial Filtering			

Package Name	Number	Mnemonic	Release
BRI line application	235	BRIL	18
— Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 233)			
— ISDN Basic Rate Interface Connected Line Presentation/Restriction			
— Bridging			
— Busy Lamp Field Array			
Business Networking Express	367	BNE	25
— Business Networking Express/EuroISDN Call Diversion			
— Business Networking Express/EuroISDN Explicit Call Transfer			
— Business Networking Express/Name and Private Number Display			
Busy Tone Detection	294	BTD	21
— China Phase II – Busy Tone Detection			
— Busy Tone Detection for Asia Pacific and CALA			
— Call Capacity Report			
Call Center Transfer Connect	393	UUI	3.0
— Call Center Transfer Connect			
Call Detail Recording Enhancement	259	CDRX	20
— Call Detail Recording Enhancement			
Call Detail Recording Expansion (7 digit)	151	CDRE	13
— Call Detail Recording Expansion			
Call Detail Recording on Teletype Terminal	5	CTY	1
— CDR on TTY			
Call Detail Recording Queue Record	83	CDRQ	3
— ACD CDR Queue Record			

Package Name	Number	Mnemonic	Release
Call Detail Recording, Data Link	6	CLNK	1
Call Detail Recording	4	CDR	1
— Call Detail Recording			
— Call Detail Recording Enhancement			
— Call Detail Recording on Redirected Incoming Calls			
— Call Detail Recording with Optional Digit Suppression			
— Call Detail Recording 100 Hour Call			
— NPI and TON in CDR Tickets			
— Call Forward and Busy Status			
— Call Forward Busy			
— Call Forward by Call Type			
— Call Forward External Deny			
— Call Forward No Answer, Second Level			
— Call Forward No Answer/Flexible Call Forward No Answer			
— Call Forward Save on SYSLOAD			
— Call Forward Save on SYSLOAD			
— Call Forward to Trunk Restriction			
— Call Forward, Break-In & Hunt Internal/External Network Wide			
— Call Forward, Internal Calls			
Call ID (for AML applications)	247	CALL ID	19
— Call Identification			
Call Page Networkwide	307	PAGENET	22
— Call Page Network Wide			

Package Name	Number	Mnemonic	Release
Call Park Networkwide — Call Park Network Wide	306	CPRKNET	22
Call Park — Call Park — Recall after Parking — Call Pickup	33	CPRK	2
Call Processor Input/Output (Option 81) — Call Processor Input/Output) — Call Redirection by Time of Day — Call Transfer	298	CPIO	21
Call Waiting Notification (Meridian 911) — Call Waiting Notification (Meridian 911) — Call Waiting/Internal Call Waiting	225	CWNT	19
Call-by-Call Service — Call-by-Call Service	117	CBC	13
Called Party Control on Internal Calls — China Phase III - Called Party Control on Internal Calls — Called Party Disconnect Control	310	CPCI	22
Calling line Identification in Call Detail Recording — Calling Line Identification in Call Detail Recording	118	CCDR	13
Calling Party Name Display — Call Party Name Display — DNIS Name Display (see also packages 98, and 113) — Calling Party Name Display Denied	95	CPND	10

Package Name	Number	Mnemonic	Release
Calling Party Privacy	301	CPP	21
— Calling Party Privacy			
— Camp-On			
— Camp-On			
— Camp-on to Multiple Appearance Directory Number			
— Capacity Expansion			
— Card LED Status			
Centralized Attendant Services (Main)	26	CASM	1
— Centralized Attendant Services - Main			
Centralized Attendant Services (Remote)	27	CASR	1
— Centralized Attendant Services – Remote			
— Centralized Multiple Line Emulation			
Charge Account for CDR	23	CHG	1
— Charge Account and Calling Party Number			
Charge Account/Authorization Code	24	CAB	1
— Charge Account/Authorization Code Base			
— Charge Display at End of Call (see also package 101)			
China Attendant Monitor Package	285	CHINA	21
— China – Attendant Monitor			
— China Number 1 Signaling – Toll Operator Break-In (see also Package 127)			
— China Number 1 Signaling Enhancements			
— China Number 1 Signaling Trunk Enhancements (see also packages 49, 113, and 128)			

Package Name	Number	Mnemonic	Release
China Toll Package	292	CHTL	21
— China Phase II – Toll Call Loss Plan			
CLASS Calling Name Delivery	333	CNAME	23
— CLASS			
CLASS Calling Number Delivery	332	CNUMB	23
— CLASS			
Collect Call Blocking	290	CCB	21
— Collect Call Blocking			
Command Status Link	77	CSL	8
— Command Status Link			
Commonwealth of Independent States Multifrequency Shuttle Signaling	326	CISMFS	23
— CIS Multifrequency Shuttle Signaling			
Commonwealth of Independent States Trunks	221	CIST	21
— Commonwealth of Independent States Digital Trunk Interface			
— Three-Wire Analog Trunk – CIS			24
— Commonwealth of Independent States Automatic Number Identification (ANI) Digits Manipulation and Gateways Enhancements			
— Commonwealth of Independent States Automatic Number Identification (ANI) Reception			24
— Commonwealth of Independent States Toll Dial Tone Detection			24
— Conference			
— Conference Warning Tone Enhancement for Italy			
Console Operations	169	COOP	14
— Console Operations			

Package Name	Number	Mnemonic	Release
Console Presentation Group	172	CPGS	15
— Console Presentation Group Level Services			
Controlled Class Of Service	81	CCOS	7
— Controlled Class of Service			
Coordinated Dialing Plan	59	CDP	1
— Coordinated Dialing Plan			
Core Network Module	299	CORENET	21
— Core Network Module			
— CP3			
Corporate Directory	381	CDIR	25
— Corporate Directory			
CSL with Alpha Signalling	85	CSLA	8
Customer Controlled Routing	215	CCR	17
— Customer Controlled Routing			
— MFC Interworking with AML Based Applications (see also packages 128, and 214)			
— Dataport Hunting			
CP Pentium® Backplane for Intel® Machine	368	CPP_CNI	25
Deluxe Hold	71	DHLD	4
— Call Hold, Deluxe			
— Call Hold, Individual Hold Enhancement			
Departmental Listed Directory Number	76	DLDN	5
Dial Intercom	21	DI	1
— Dial Intercom			
— Distinctive Ringing for Dial Intercom			
— Dial Pulse/Dual-tone Multifrequency Conversion			

Package Name	Number	Mnemonic	Release
Dial Tone Detector	138	DTD	10
— Dial Tone Detection			
— Flexible Dial Tone Detection			
Dialed Number Identification System	98	DNIS	10
— Dialed Number Identification Services			
— Dialed Number Identification Services Length Flexibility			
— Dialed Number Identification Services Name Display (see also packages 95, and 131)			
— 7 Digit DNIS for MAX			
— N Digit DNIS			24
Digit Display	19	DDSP	1
— Digit Display			
Digit Key Signaling	180	DKS	1
Digital Access Signaling System 2	124	DASS2	16
— Analog Private Network Signaling System (APNSS) (see also packages 190, 122, and 123)			
— DASS2/DPNSS1 – Integrated Digital Access (see also packages 122, and 123)			

Package Name	Number	Mnemonic	Release
Digital Private Network Signaling Network Services (DPNSS1) — Attendant Call Offer — Attendant Timed Reminder Recall and Attendant Third Party Service — Call Back when Free and Next Used — D-channel Handler Interface Expansion — Extension Three-Party Service — Loop Avoidance — Redirection — Route Optimization — Step Back on Congestion — Diversion — Night Service — Route Optimisation/MCDN Trunk Anti-Tromboning Interworking	231	DNWK	16
Digital Private Network Signaling System 1 Message Waiting Indication — DPNSS1 Message Waiting Indication	325	DMWI	23
Digital Private Network Signaling System 1 — Analog Private Network Signaling System (APNSS) (see also packages 190, 122, and 124) — DASS2/DPNSS1 – Integrated Digital Access (see also packages 122, and 124) — Digital Trunk Interface Enhancements — Digitone Receiver Enhancements: – Digitone Receiver Time-out Enhancement — Digitone Receiver Enhancements: – Quad Density Digitone Receiver Card	123	DPNSS	16
Direct Inward Dialing to TIE (Japan only) — Direct Inward Dialing to TIE	176	DTOT	16

Package Name	Number	Mnemonic	Release
— Direct Inward Dialing to TIE Connection			
Direct Inward System Access	22	DISA	1
— Call Park on Unsupervised Trunks			
— Direct Inward System Access			
— Direct Inward System Access on Unsupervised Trunks			
Direct Private Network Access	250	DPNA	21
— Direct Private Network Access			
Directed Call Pickup	115	DCP	12
— Call Pickup, Directed			
— Directory Number Delayed Ringing			
Directory Number Expansion (7 Digit)	150	DNXP	13
— Directory Number Expansion			
— Directory Number • Flexible Attendant Directory Number • Listed Directory Numbers • Single Appearance Directory Number • Multiple Appearance Directory Number • Prime Directory Number			
— Diskette Overflow Warning			
— Display of Calling Party Denied			
Distinctive Ringing	74	DRNG	4/9
— Distinctive/New Distinctive Ringing			
Do Not Disturb, Group	16	DNDG	1
— Do Not Disturb Group			

Package Name	Number	Mnemonic	Release
Do Not Disturb, Individual	9	DNDI	1
— Do Not Disturb			
— Network Individual Do Not Disturb (see also packages 127, and 159)			
— Electronic Brandlining			
Emergency Services Access Calling Number Mapping	331	ESA_CLMP	23
— Emergency Services Access (See also packages 329 and 330)			
Emergency Services Access Supplementary	330	ESA_SUPP	23
— Emergency Services Access (See also packages 329 and 331)			
Emergency Services Access	329	ESA	23
— Emergency Services Access (See also packages 330 and 331)			
— End of Selection			
— End of Selection Busy			
— End-of-Dialing on Direct Inward/Outward Dialing Incoming Call Indicator Enhancement			
End-To-End Signaling	10	EES	1
— Attendant End-to-End Signaling			
— End-to-End Signaling			
Enhanced ACD Routing	214	EAR	17
— Enhanced Automatic Call Distribution Routing			
— MFC Interworking with AML Based Applications (see also packages 128, and 215)			
Enhanced Call Trace	215	ECT	18
— Customer Controlled Routing			
— MFC Interworking with AML Based Applications (see also packages 128, and 214)			

Package Name	Number	Mnemonic	Release
Enhanced Controlled Class of Service	173	ECCS	15
Enhanced DPNSS Services — DPNSS1 Executive Intrusion	288	DPNSS_ES	21
Enhanced DPNSS1 Gateway — Enhanced DPNSS1 Gateway	284	DPNSS189I	20
Enhanced Hot Line — Hot Line — Network Intercom — Enhanced input/output buffering — Enhanced Maintenance (Patching)	70	HOT	4/10
Enhanced Music — Music, Enhanced	119	EMUS	12
Enhanced Night Service — Enhanced Night Service — Enhanced package printout — Equal Access Compliance	133	ENS	20
Euro ISDN Trunk - Network Side — EuroISDN Trunk - Network Side	309	MASTER	22
Euro ISDN — ISDN – Advice of Charge for EuroISDN — ISDN BRI and PRI Trunk Access for Europe (EuroISDN) — EUROISDN Continuation	261	EURO	20
Euro Supplementary Service — EuroISDN Call Completion Supplementary Service	323	ETSI_SS	22

Package Name	Number	Mnemonic	Release
Executive Distinctive Ringing	185	EDRG	16
— Executive Distinctive Ringing			
Fast Tone and Digit Switch	87	FTDS	7
— Fast Tone Digit Switch			
FCC Compliance for DID Answer Supervision	223	FCC68	17
— Federal Communications Commission Compliance for DID Answer Supervision			
Feature Group D	158	FGD	17
— Feature Group D (Inbound to Meridian 1)			
— Federal Communications Commission Compliance for Equal Access			
— First-Second Degree Busy Indication			
— First-Second Degree Busy Indication, ISDN			
— Flexible Attendant Call Waiting Thresholds			
— Flexible Busy Tone Timer			
Fiber Network	365	FIBN	25
Flexible Call Back Queuing	61	FCBQ	1
— Flexible Call Back Queuing			
Flexible Direct Inward Dialing	362	FDID	24
— Flexible Direct Inward Dialing			

Package Name	Number	Mnemonic	Release
Flexible Feature Codes	139	FFC	15
— Call Forward/Hunt Override Via Flexible Feature Code			
— China Number 1 Signaling – Flexible Feature Codes			
— Dial Access to Group Calls (see also package 48).			
— Direct Inward Dialing Call Forward No Answer Timer			
— Electronic Lock Network Wide/Electronic Lock on Private Lines			
— Flexible Feature Codes			
— Automatic Wake FFC Delimiter			
— Call Forward Destination Deactivation			
— Flexible Key Assignment			
Flexible Numbering Plan	160	FNP	14
— Alternative Routing for DID/DOD			
— Flexible Numbering Plan			
— Special Dial Tones after Dialed Numbers			
— Flexible Numbering Plan Enhancement			
— Flexible Orbiting Prevention Timer			
Flexible Tones and Cadences	125	FTC	16
— Flexible Tone and Digit Switch Control			
— Reverse Dial on Routes and Telephones			
— Tones and Cadences			
Forced Charge Account	52	FCA	1
— Charge Account, Forced			

Package Name	Number	Mnemonic	Release
French Type Approval	197	FRTA	15
— Camp-on to a Set in Ringback or Dialing			
— Forward No Answer Call Waiting Direct Inward Dialing			
— Group Hunt Queuing (see also package 120)			
— Group Hunt Queuing Limitation Enhancement (see also package 120)			
— Loopback on Central Office Trunks			
Group Call	48	GRP	1
— Dial Access to Group Calls (see also package 139).			
— Group Call			
— Group Hunt Queuing Limitation (see also package 120)			
Group Hunt/DN Access to SCL	120	PLDN	15
— Group Hunt Queuing (see also package 197)			
— Group Hunt Queuing Limitation (see also package 131)			
— Group Hunt Queuing Limitation Enhancement (see also package 197)			
— Group Hunt			
— Speed Call Directory Number Access			
— Handset Volume Reset			
— Handsfree Download (Meridian Digital Telephones			
— Held Call Clearing			
H323 Virtual Trunk	399	H323_VTRK	3.0
— IP Peer Networking Phase 2			
— Succession Branch Office			
HiMail Fax Server	195	FAXS	18

Package Name	Number	Mnemonic	Release
History File	55	HIST	1
— History File			
Hold in Queue for IVR	218	IVR	18
Hospitality Management	166	HOSP	16
Hospitality Screen Enhancement	208	HSE	17
— Hospitality Enhancements: Display Enhancements			
— Hunting By Call Type			
— Hunting • Circular Hunting • Linear Hunting • Secretarial Hunting • Short Hunting • Data Port Hunting • Trunk Hunting			
— Incoming Call Indicator Enhancement			
Incoming DID Digit Conversion	113	IDC	12
— China Number 1 Signaling Trunk Enhancements (see also packages 49, 128, and 131)			
— DNIS Name Display (see also packages 95, and 98)			
— Incoming DID Digit Conversion			
— Incoming Trunk Programmable Calling Line Identification			
— Incremental Software Management			
— Input/Output Access and System Limits			

Package Name	Number	Mnemonic	Release
Integrated Digital Access — Analog Private Network Signaling System (APNSS) (see also packages 190, 123, and 124) — DASS2/DPNSS1 – Integrated Digital Access (see also packages 123 and 124) — DPNSS1 Satellite — DASS2/DPNSS INIT Call Cutoff	122	IDA	16
Integrated Message System **UST and UMG are part of IMS Package.** — Integrated Messaging System Link	35	IMS	2
Integrated Services Digital Network Application Module Link for Third Party Vendors — Application Module Link — Network Application Protocol Link Enhancement	153	IAP3P	13
Integrated Services Digital Network BRI Trunk Access — Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 235)	233	BRIT	18
Integrated Services Digital Network Supplementary Features — Call Connection Restriction (see also packages 146 and 147) — Direct Inward Dialing to Network Calling — Incoming Digit Conversion Enhancement — Network Time Synchronization — X08 to X11 Gateway	161	ISDN INTL-SUP	14
Integrated Services Digital Network Signaling Link — Call Connection Restriction (see also packages 146 and 161)	147	ISL	13

Package Name	Number	Mnemonic	Release
Integrated Services Digital Network	145	ISDN	13
— Backup D-Channel to DMS-100/250 and AT&T 4ESS			
— Call Pickup Network Wide			
— D-Channel Error Reporting and Monitoring			
— Integrated Services Digital Network (ISDN) Primary Rate Interface			
— Network Name Display (Meridian 1 to DMS-100/250)			
— Total Redirection Count			
— T309 Time			
— Integrated Voice and Data			
Intercept Computer Interface	143	ICP	10
— Intercept Computer Dial from Directory			
— Intercept Computer Enhancements			
— Intercept Computer Flexible DN Length			
— Intercept Computer Interface			
— Intercept Computer Meridian Mail Interactions			
— Intercept Computer Network Screen Activation, Flexible DN, Meridian Mail Interactions			
— Intercept Treatment Enhancements			
Intercept Treatment	11	INTR	1
— Intercept Treatment			
Inter-Exchange Carrier	149	IEC	13
— Inter Exchange Carrier			
Internal CDR	108	ICDR	10
— Internal Call Detail Recording			
International 1.5/2.0 Mb/s Gateway	167	GPRI	18
— Radio Paging			

Package Name	Number	Mnemonic	Release
— International Meridian 1			
International nB+D	255	INBD	20
— ISDN PRI D70 Trunk Access for Japan (nB+D)			
International Primary Rate Access (CO)	146	PRA	13
— Call Connection Restriction (see also packages 147 and 161)			
— Integrated Services Digital Network Primary Rate Access			
— Integrated Services Digital Network Primary Rate Access Central Office Connectivity to Japan D70			
International Primary Rate Access	202	IPRA	15
— Integrated Services Access/Call by Call Service Selection Enhancements			
— Integrated Services Digital Network Primary Rate Access to 1TR6 Connectivity			
— Integrated Services Digital Network Primary Rate Access to NUMERIS Connectivity			
— Integrated Services Digital Network Primary Rate Access to SwissNet 2 Connectivity			
— Integrated Services Digital Network Primary Rate Access to SYS-12 Connectivity			
International Supplementary Features	131	SUPP	9
— IODU/C			
IP Expansion	295	IPEX	25.40
— IP Expansion			
ISDN Semi-Permanent Connection	313	ISPC	22
— ISDN Semi-Permanent Connections for Australia			
— Italian Central Office Special Services (see also packages 129, and 157)			

Package Name	Number	Mnemonic	Release
Japan Central Office Trunks	97	JPN	9
— Japan Central Office Trunk			
Japan Digital Multiplex Interface	136	JDMI	14
— Japan Digital Multiplex Interface			
Japan Telecommunication Technology Committee	335	JTTC	23
— Japan TTC Common Channel Signaling			
Japan Tone and Digit Switch	171	JTDS	14
— Japan Tone and Digit Switch			
Last Number Redial	90	LNR	8
— Last Number Redial			
Latin American Spanish	279	MLMS_SPL	20
— Latin American Spanish			
Limited Access to Overlays	164	LAPW	16
— B34 Dynamic Loss Switching (see also packages 131 and 203)			
— Faster I/O			
— Limited Access to Overlays			
— Limited Access to Overlays Password Enhancement			
— Teletype Terminal Access Control in Multi-Customer Environment (see also package 131)			
Line Load Control	105	LLC	10
— Line Load Control			
— Line Lockout			
Local Steering Code Modifications	137	LSCM	10
— Local Steering Code Modifications			
— Lockout, DID Second Degree Busy and MFE Signaling Treatments			

Package Name	Number	Mnemonic	Release
— LOGIVOX Telephone			
— Loop Start Answer Supervision XUT			
— Loop Start Supervisory Trunks			
— Loop Start Supervisory Trunks (Incoming Calls)			
M2000 Digital Sets — Distinctive Ringing for Digital Telephones — M2312 Digit Display — M2317 Telephones — Flexible Voice/Data Terminal Number	88	DSET	7
M2250 Attendant Console — Digital Attendant Console	140	DCON	15
M2317 Digital Sets — M2317 Digital Sets	91	DLT2	9
M3000 Digital Sets — M3000 Telephones	89	TSET	7
M3900 Full Icon Support — M3900 Full Icon Support	397	ICON_ PACKAGE	3.0
M3900 Phase III Virtual Office Enhancement — Virtual Office Enhancement	387	VIR_OFF_ ENH	25.40
M3900 Ring Again	396	M3900_RGA_ PROG	3.0
M911 Enhancement Display — 10/20 Digit ANI on 911 Calls	249	M911 ENH	25
Maid Identification — Maid Identification	210	MAID	17
— Make Set Busy and Voice Call Override			

Package Name	Number	Mnemonic	Release
Make Set Busy	17	MSB	1
— Make Set Busy			
— Make Set Busy Improvement			
— Malicious Call Trace on Direct Inward Dialing			
Malicious Call Trace	107	MCT	10
— Enhanced Malicious Call Trace			
— Malicious Call Trace			
— Malicious Call Trace DN/TN Print			
— Malicious Call Trace Idle			
— Manual Line Service			
— Manual Service Recall to Attendant			
— Manual Signaling (Buzz)			
— Manual Trunk Service			
MAT 5.0	296	MAT	22
— Meridian 1 Attendant Console Enhancements (see also package 76)			
Meridian 1 Companion Option	240	MCMO	19
— Meridian 1 COMPANION			
— Meridian Companion Enhanced Capacity			24
MCDN End to End Transparency	348	MEET	24
Meridian 1 Enhanced Conference, TDS and MFS	204	XCT0	15
— Meridian 1 Enhanced Conference, TDS and MFS			
Meridian 1 Fault Management	243	ALRM_FILTER	19
— Alarm Management			
— Meridian 1 Initialization Prevention and Recovery			
Meridian 1 Microcellular Option	303	MMO	22

Package Name	Number	Mnemonic	Release
Meridian 1 Mobility Multi-Site Networking	314	MMSN	22
Meridian 1 Packet Handler	248	MPH	19
— Meridian 1 Packet Handler			
Meridian 1 Superloop Administration (LD 97)	205	XCT1	15
— Extended DID/DOD Software Support – Europe			
— Extended Flexible Central Office Trunk Software Support			
— Extended Tone Detector and Global Parameters Download (see also package 203)			
— Generic XFCOT Software Support			
Meridian 1 XPE	203	XPE	15
— B34 Codec Static Loss Plan Downloading			
— B34 Dynamic Loss Switching (see also packages 131, and 164)			
— Extended Multifrequency Compelled Sender/ Receiver			
— Extended Tone Detector and Global Parameters Download (see also package 205)			
— Intelligent Peripheral Equipment Software Support Enhancements			
Meridian 911	224	M911	19
— Meridian 911 Enhancements – Call Abandon			
— Meridian 911 Enhancements – MADN Display Coordination			
Meridian Hospitality Voice Service	179	HVS	16
— Meridian Hospitality Voice Services			
Meridian Link Modular Server	209	MLM	16
— Meridian Link Enhancements			
Meridian SL-1 ST Package	96	SLST	9
— Meridian SL-1 ST Package			

Package Name	Number	Mnemonic	Release
Message Intercept	163	MINT	15
— Message Intercept			
Message Waiting Center	46	MWC	1
— Message Waiting Lamp Maintenance			
— Message Waiting Unconditional			
— Meridian Mail Trunk Access Restriction			
Message Waiting Indication Interworking with DMS	219	MWI	19
— Message Waiting Indication (MWI) Interworking			
Mini CDR	31	MCDR	1
Mobility Server	302	MOSR	22
— Modular Telephone Relocation			

Package Name	Number	Mnemonic	Release
Multifrequency Compelled Signaling	128	MFC	9
— China Number 1 Signaling Trunk Enhancements (see also packages 49, 113, and 131)			
— China Number 1 Signaling – Active Feature Dial Tone (see also package 126)			
— China Number 1 Signaling – Audible Alarm (see also package 126)			
— China Number 1 Signaling – Vacant Number Announcement (see also package 126)			
— India Phase 2			
— R2 Multifrequency Compelled Signaling (MFC) DID/DTMF DOD			
— R2 Multifrequency Compelled Signaling (MFC) Selective Route To Attendant			
— MFC Interworking with AML Based Applications (see also packages 214 and 215)			
— R2Multifrequency Compelled Signaling Timer Control			
— Semi-Compelled MFC and Calling Name Identification Charges			
Multifrequency Signaling for Socotel	135	MFE	10
— Multifrequency Signaling for Socotel			
Multi-Language I/O Package	211	MLIO	16
— Multi-language TTY Input/Output			
Multi-Language Wake Up	206	MLWU	16
— Multi-language Wake Up			
— Multi-Party Operation Enhancements			
Multi-Party Operations	141	MPO	20
— Attendant Clearing during Night Service			
— Multi-Party Operations			
— Multiple Appearance DN Redirection Prime			

Package Name	Number	Mnemonic	Release
— Multiple Console Operation			
Multiple Queue Assignment	297	MQA	21
— Multiple Queue Assignment			
Multiple-Customer Operation	2	CUST	1
— Multiple Customer Operation			
Multiple-Tenant Service	86	TENS	7
— Multi-Tenant Service			
Multi-purpose Serial Data Link Serial Data Interface	227	MSDL SDI	19
— Multi-purpose Serial Data Link Serial Data Interface			
Multi-purpose Serial Data Link Single Terminal Access	228	MSDL STA	19
— Single Terminal Access			
Multi-purpose Serial Data Link	222	MSDL	18
— Multi-purpose Serial Data Link			
Multi-Site Mobility Networking	370	MSMN	25
Multi-User Login	242	MULTI_USER	19
— Multi-User Login			
Music Broadcast	328	MUSBRD	23
— Music Broadcast			
Music	44	MUS	1
— Music			
N/W Communications Management Center	30	CMAC	1

Package Name	Number	Mnemonic	Release
Network Alternate Route Selection — Equi-distribution Network Attendant Service Routing (see also package 159) — Network Alternate Route Selection/Basic Alternate Route Selection Enhancement – Local Termination (see also package 57) — Network Anti-tromboning — Virtual Network Services/Virtual Directory Number Expansion (see also package 183)	58	NARS	1
Network Attendant Service — Equi-distribution Network Attendant Service Routing (see also package 58) — Network Individual Do Not Disturb (See also packages 9 and 127).	159	NAS	20
Network Authorization Code — Network Authorization Code	63	NAUT	1
Network Automatic Call Distribution — Network Automatic Call Distribution	207	NACD	15
Network Call Back Queuing — Network Call Back Queuing	38	MCBQ	2
Network Call Transfer	67	NXFR	3
Network Class Of Service — Network Class of Service	32	NCOS	1
Network Message Services	175	NMS	16
Network Priority Queuing — Network Priority Queuing	60	PQUE	1
Network Signaling — Network Signaling	37	NSIG	2

Package Name	Number	Mnemonic	Release
Network Speed Call	39	NSC	2
— Network Speed Call			
Network Traffic Measurements	29	NTRF	1
— Network Traffic Measurement			
New Flexible Code Restriction	49	NFCR	2
— China Number 1 Signaling Trunk Enhancements (see also packages 113, 128, and 131)			
— New Flexible Code Restriction			
New Format CDR	234	FCDR	18
— Call Detail Recording Time to Answer			
— CDR on Busy Tone			
Next Generation Connectivity	324	NGEN	22
NI-2 Call By Call Service Selection	334	NI-2 CBC	23
— Night Restriction Classes of Service			
— Night Service			
— Night Service Enhancements – All Calls Remain Queued for Night Service			
— Night Service Enhancements – Recall to Night DN			
— Night Service Enhancements – Requeuing of Attendant Present Calls			
— Night Service Enhancements – Requeuing of Attendant Present Calls			
NI-2 Name Display Service	385	NDS	25.40
— NI-2 Name Display Supplementary Service			
Nortel Symposium Call Center	311	NGCC	22
North America National ISDN Class II Equipment	291	NI2	21

Package Name	Number	Mnemonic	Release
— North American Numbering Plan			
— Off-Hook Alarm Security			
Observe Agent Security	394	OAS	3.0
— Observe Agent Security			
Off-Hook Queuing	62	OHQ	1
— Network Drop Back Busy and Off-hook Queuing (see also package 192)			
Office Data Administration System	20	ODAS	1
— Office Data Administration System			
— Off-Premise Extension			
On Hold On Loudspeaker	196	OHOL	20
— On-Hook Dialing			
Open Alarms	315	OPEN ALARM	22

Package Name	Number	Mnemonic	Release
Operator Call Back (China #1)	126	OPCB	14
— Busy Verify on Calling Party Control Calls			
— China Number 1 Signaling – Active Feature Dial Tone (see also package 128)			
— China Number 1 Signaling – Audible Alarm (see also package 128)			
— China Number 1 Signaling – Called Party Control			
— China Number 1 Signaling – Calling Number Identification on Outgoing Multifrequency Compelled Signaling			
— China Number 1 Signaling – Calling Party Control			
— China Number 1 Signaling – Flexible Timers			
— China Number 1 Signaling – KE Multifrequency Compelled Tandem Signaling			
— China Number 1 Signaling – Malicious Call Trace Enhancement			
— China Number 1 Signaling – Off-hook Tone			
— China Number 1 Signaling – Toll Call Identification			
— China Number 1 Signaling – Toll Operator Call Back			
— China Number 1 Signaling – Toll Operator Call Back Enhancement			
— China Number 1 Signaling – Vacant Number Announcement (see also Package 128)			

Package Name	Number	Mnemonic	Release
Optional Features	1	OPTF	1
— Autodial			
— Call Forward All Calls			
— Ring Again			
— Speed Call			
— Speed Call on Private Lines (see also package 0)			
— Speed Call/Autodial with Authorization Codes (see also package 34)			
— Speed Call Delimiter (see also package 34)			
Optional Outpulsing Delay	79	OOD	5
— Optional Outpulsing Delay			
Originator Routing Control	192	ORC_RVQ	18
— Network Drop Back Busy and Off-hook Queuing (see also package 62)			
— Remote Virtual Queuing			
— Out-of-Service Unit			
Outpulsing, asterisk (*) and octothorpe (#)	104	OPAO	
— Outpulsing of Asterisk “*” and Octothorpe “#”			
Overlap Signaling (M1 to M1 and M1 to 1TR6 CO)	184	OVLP	15
— Overlap Signaling			
— Overlay 45 Limited Repeats			
— Overlay Cache Memory			
— Override			
— Paging			
— Partial Dial Timing			
— PBX (500/2500) Telephones			
— Periodic Camp-on Tone			

Package Name	Number	Mnemonic	Release
— Periodic Clearing			
— Periodic Clearing Enhancement			
— Periodic Clearing on RAN, Meridian Mail, ACD, and Music			
Personal Call Assistant	398	PCA	3.0
— Personal Call Assistant			
Phantom TN	254	PHTN	20
— Phantom TNs			
— Position Busy with Call on Hold			
PPM/Message Registration	101	MR	10
— Advice of Charge Real-time Supplementary Services for NUMERIS and SWISSNET (see also package 131)			
— Advice of Charge – Charging Information and End of Call for NUMERIS Connectivity (see also package 131)			
— Message Registration			
— Periodic Pulse Metering			
— Predictive Dialing			
Pretranslation	92	PXLT	8
— Pretranslation			
— Preventing Reciprocal Call Forward			
Priority Network Override	389	PONW	25.40
— Network Breakin and Force Disconnect			
Priority Override/Forced Camp-On	186	POVR	20
— Forced Camp-on and Priority Override			
— Privacy			
— Privacy Override			

Package Name	Number	Mnemonic	Release
— Privacy Release			
— Private Line Service			
Property Management System Interface	103	PMSI	10
— Property Management System Interface			
— Public Switched Data Service			
Pulsed E&M (Indonesia, French Colisée)	232	PEMD	18
— Pulsed E&M DTI2 Signaling			
Q Reference Signaling Point Interface	263	QSIG	20
— Integrated Services Digital Network QSIG Basic Call			
— QCW and M1250 Attendant Console Capabilities			
QSIG Generic Functional protocol	305	QSIG GF	22
— ISDN QSIG Generic Functional Transport			
QSIG Supplementary Service	316	QSIG-SS	22
— ISDN QSIG Call Completion			
— ISDN QSIG Call Diversion Notification			
— ISDN QSIG Path Replacement			
Radio Paging	187	RPA	15
— Radio Paging			
— Radio Paging Product Improvements			
— Recall to Same Attendant			
— Recall to Same Attendant			
— Recall with Priority during Night Service			
— Recall With Priority during Night Service			
— Recall With Priority during Night Service Network Wide			

Package Name	Number	Mnemonic	Release
Recorded Announcement Broadcast	327	RANBRD	23
— Recorded Announcement Broadcast			
Recorded Announcement	7	RAN	1
— Recorded Announcement			
Recorded Overflow Announcement	36	ROA	2
— Recorded Overflow Announcement			
— Recorded Telephone Dictation			
— Recovery of Misoperation on the Attendant Console			
— Recovery on Misoperation of Attendant Console			
— Reference Clock Switching			
— Reference Clock Switching (see also packages 75, 129, and 154)			
Remote IPE	286	REMOTE_IPE	
— Remote Intelligent Peripheral Equipment			
Remote Virtual Queuing	192	RVQ	18
— Network Drop Back Busy and Off-hook Queuing (see also package 62)			
— Remote Virtual Queuing			
Resident Debug	82	RSDB	9
— Restricted Call Transfer			
— Restricted Direct Inward Dialing Class of Service			
— Ring Again on No Answer (see also package 148)PBX (500/2500) Telephones			
— 500/2500 Line Disconnect			
— Ring and Hold Lamp Status			
— Ringback Tone from Meridian 1 Enhancement			

Package Name	Number	Mnemonic	Release
Ringing Change Key	193	RCK	15
— Ringing Change Key			
Room Status	100	RMS	10
— Room Status			
Scheduled Access Restrictions	162	SAR	20
— Scheduled Access Restrictions			
— Secrecy Enhancement			
— Secretarial Filtering			
— Seizure Acknowledgment			
— Selectable Conferee Display and Disconnect			
— Selectable Directory Number Size			
Semi-Automatic Camp-On	181	SACP	15
— Attendant Blocking of Directory Number			
— Attendant Idle Extension Notification			
— Semi-Automatic Camp-On			
— Serial Port Expansion			
Series Call	191	SECL	15
— Series Call			
Set Relocation	53	SR	1
— Automatic Set Relocation			
— Short Buzz for Digital Telephones			
— Short Memory Test			
— Single Digit Access to Hotel Services			
Set-to-Set Messaging	380	STS	25
— Set-to-Set Messaging			

Package Name	Number	Mnemonic	Release
Single Term Access	228	STA	19
— Single Term Access			
— SL-1 Telephones			
— Slow Answer Recall Enhancement			
— Slow Answer Recall for Transferred External Trunks			
— Source Included when Attendant Dials			
Spanish KD3 DID/DOD interface	252	KD3	20
— KD3 Direct Inward Dialing/Direct Outward Dialing for Spain			
— Special Signaling Protocols			
— Special Trunk Support			
— Speed Call Directory Number Access			
— Speed Call on Private Lines (see also package 1)			
— Speed-Up Data Dump			
Standalone Meridian Mail	262	SAMM	20
— Meridian Mail, Standalone			
Station Activity Records	251	SCDR	20
— Station Activity Records			
Station Camp-On	121	SCMP	20
— Station Camp-On			
Station Category Indication	80	SCI	7
— Station Category Indication			
Station Loop Preemption	106	SLP	10
Station Specific Authorization Codes	229	SSAU	19
— Station Specific Authorization Code			

Package Name	Number	Mnemonic	Release
— Station-to-Station Calling			
Stored Number Redial	64	SNR	3
— Stored Number Redial			
Supervisory Attendant Console	93	SUPV	8
— Supervisory Attendant Console			
Supervisory Console Tones	189	SVCT	20
— System Capacity Enhancements			
System Errors and Events Lookup	245	SYS_MSG_ LKUP	19
— System Message Lookup			
System Speed Call	34	SSC	2
— Speed Call/Autodial with Authorization Codes (see also package 1)			
— Speed Call, System			
— Speed Call Delimiter (see also package 34)			
— Telephones (PBX and SL-1)			
— Teletype Terminal Access Control in Multi-Customer Environment (see also package 164)			
— Telset Call Timer Enhancement			
Time and Date	8	TAD	1
— Time and Date			
Tone Detector Special Common Carrier	66	SCC	7
Tone Detector	65	TDET	7
— Tone Detector			
— Tone to Last Party			
— Tones, Flexible Incoming			
Traffic Monitoring	168	TMON	

Package Name	Number	Mnemonic	Release
Trunk Anti-Tromboning	293	TAT	21
— Trunk Anti-Tromboning			
Trunk Barring	132	TBAR	20
— Trunk Barring			
Trunk Failure Monitor	182	TFM	15
— Trunk Failure Monitor			
— Trunk Failure Monitor Enhancement			
Trunk Hook Flash (Centrex)	157	THF	14
— Centrex Switchhook Flash			
— Italian Central Office Special Services (see also packages 129, and 131)			
— Trunk to Trunk Connections			
— Trunk Traffic Reporting Enhancement			
Trunk Verification from Station	110	TVS	9.32
— Trunk Verification from a Station			
— Uninterrupted Line Connection			
United Kingdom	190	UK	16
— Analog Private Network Signaling System (APNSS) (see also packages 122 123, and 124)			
— UK Analogue Hardware Support			
Universal ISDN Gateways	283	UIGW	20
— Universal ISDN Gateway			
— Variable Flash Timing and Ground Button			
— Variable Guard Timing			
VIP Auto Wake Up	212	VAWU	17
— Hospitality Enhancements: V.I.P. Auto Wake Up			

Package Name	Number	Mnemonic	Release
Virtual Network Services — Virtual Network Services — Virtual Network Services/Virtual Directory Number Expansion (see also package 58) — Voice Call	183	VNS	16
Virtual Office — Emergency Services For Virtual Office — Internet Telephone Virtual Office — Succession Branch Office — Virtual Office	382	VIRTUAL_ OFFICE	25
Virtual Office Enhancement — Emergency Services For Virtual Office — Internet Telephone Virtual Office — Succession Branch Office	387	VOE	3.0
Voice Mailbox Administration — Meridian Mail Voice Mailbox Administration	246	VMBA	19
X08 to X11 Gateway — X08 to X11 Gateway	188	L1MF	15

10/20 Digit ANI on 911 Calls

Contents

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Feature description

This feature brings the system into compliance with the Federal Communications Commission (FCC) decision that requires a circuit switched network, working as a Public Safety Answering Point (PSAP), to accept a 10 or 20 digit Automatic Number Identification (ANI) when terminating 911 calls.

10 digit ANI feature

The 10 digit ANI feature addresses the increasing number of Numbering Plan Areas (NPAs) in North America. The increasing number of NPAs requires that a single PSAP must be capable of handling multiple NPAs within its jurisdiction.

The 10/20 Digit ANI on 911 Calls feature changes the ANI format to include the NPA in the ANI field. A single PSAP can handle any number of valid NPAs with the 10 digit format.

20 digit ANI feature

The 20 digit ANI feature addresses the problem of accurately determining the location of a wireless calling party dialing 911.

The first 10 ANI digits provide the Calling Station Number (CSN). The CSN for a 911 call is the Calling Party Number (CPN), if available, or the billing number if the CPN is not available. The CPN, if available, is used to call the originator back when a 911 call is disconnected.

The second 10 ANI digits, or Pseudo Automatic Number Identification (PANI), provides the cell site and sector information to best define the wireless calling party's location. The PANI allows emergency assistance to be sent to the correct area.

II digit definition

The 10/20 Digit ANI feature replaces the NPD with two II digits. The definition of II digits is as follows:

- 40 for normal display
- 44 for flashing display (Default Routing)
- 48 for a test call

Note: The system uses an attached "*" instead of a "flashing display". Default Routing is used when the Selective Routing process at the Central Office does not produce a valid Emergency Service Number (ESN). If no valid CSN information is available on a wireline call, or if no valid cell site and sector information is available on a wireless call, the call is sent to the default ESN associated with the incoming trunk group for that call.

CSN wireline calls format

The CSN wireline call format is as follows:

KP II NPA NXX YYYY STP

Where:

- KP is the key pulse.
- NPA NXX YYYY represents the originator's CSN.
- STP is a digit that tells the system that there is only 10 digits. Termination of the call occurs immediately after receiving the STP digit.

CSN wireless calls format

The CSN wireless call format is as follows:

KP II NPA NXX YYYY ST KP X...X ST

Where:

- KP is the key pulse.
- NPA NXX YYYY represents the originator's CSN.
- The first ST digit flags the call register as a wireless call for display purposes.
- The second KP marks the beginning of the PANI.
- X...X represents the cell site and sector identification. Although 10 digits are required for this information to be complete, any available information is sent. Therefore, this information can range from 0 to 10 digits.
- The second ST digit terminates the call.

Digit Display

Wireline

Wireline M911 calls display on a digital set as follows:

- For calls with II digits equal to 40, the 10 digits display as:
  - NPA NXX YYYY
- For calls with II digits equal to 44, the 10 digits display as:
  - NPA NXX YYYY*

Wireless

Wireless M911 calls display on a digital set as follows:

- For calls with II digits equal to 40, the 20 digits display as:
  - (PANI) NPA NXXX YYYY WIRELESS
  - (CSN) NPA NXX YYYY
- For calls with II digits equal to 44, the 20 digits display as:
  - (PANI) NPA NXXX YYYY WIRELESS
  - (CSN) NPA NXX YYYY*

911E (end-office) call processing

With the 10/20 digit ANI for 911 Calls feature, the system continues to expect the dialed digit(s) first.

The dialed digit format is KP+digits+ST, where the digit(s) are 911, 11, or 1, followed by the ANI CSN information.

911T (tandem) call processing

With the 10/20 digit ANI for 911 Calls feature, the system does not expect the dialed digit(s) (911, 11, or 1), only the ANI CSN information.

Operating parameters

This feature is compatible with the system.

The functionality of the 10/20 Digit ANI on 911 Calls feature depends on the local telephone company to comply with Bellcore GR-2953. Therefore, the ability to collect the 10/20 digit ANI formats must be enabled on a separate trunk route basis.

If the 20 digit wireless calls are tandem to the ISDN route, the display shows the II + 10 digit CSN.

The Custom Local Area Signaling Service (CLASS) set only displays up to 10 digit ANI.

Feature interactions

Call Trace

Call Trace in LD 80 is modified to show II NPID + 10 digit ANI information. The Call Trace record also shows the PANI information.

Call Detail Recording

The Call Detail Recording record (with package 234) is modified to display PANI for wireless calls when FCDR = NEW in LD 17.

Display on CLASS sets

Only 10 digit ANI will display on class set for both 911E or 911T trunk. The PANI will not display. However, if it is a wireless call, the PANI can be traced by LD 80.

Display on tandem call

Only II + 10 digit ANI will display on the telephone set when M911 calls are forwarded or transferred through ISDN or PRA routes. This only applies for 911E route types.

Malicious Call Trace

The Malicious Call Trace record is modified to show II NPID + 10 digit ANI information. The record also contains the PANI information.

Feature packaging

M911 Enhancement Display (M911 ENH) package 249 is introduced with this feature.

The 10/20 Digit ANI on 911 Calls feature requires the following packages:

- Digit Display (DDSP) package 19
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package B (ACDB) package 41
- Automatic Call Distribution Package A (ACDA) package 45

- Enhanced Automatic Call Distribution Routing (EAR) package 214
- Meridian 911 (M911) package 224
- Call Waiting Notification (CWNT) package 225
- M911 Enhancement Display (M911 ENH) package 249

The following additional packages are not required, but are recommended:

- At least one of either Call Detail Recording (CDR) package 4 or Call Detail Recording on Teletype Machine (CTY) package 5
- Automatic Call Distribution Package C (ACDC) package 42

  Note: package 42 is not needed if packages 51 and 52 are enabled
- Automatic Call Distribution Load Management Reports (LMAN) package 43
- Automatic Call Distribution Package D (ACDD) package 50
- Automatic Call Distribution Package D, Auxiliary Link Processor (LNK) package 51
- Call Party Name Display (CPND) package 95
- Malicious Call Trace (MCT) package 107
- Calling Line Identification in Call Detail Recording (CCDR) package 118
- Flexible Tones and Cadences (FTC) package 125
- Limited Access to Overlays (LAPW) package 164
- New Format CDR (FCDR) package 234 (recommended for wireless calls)

  Note: The M911 Call Abandon feature is included in Meridian 911 (M911) package 224, and requires Call Identification (CALL ID) package 247. If an application also requires Meridian Link, Meridian Link Module (MLM) package 209 is required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Set the Pseudo Automatic Number Identification (PANI) prompt to YES to display PANI.

2 LD 16 – Configure the M911 ANI format.

LD 15 – Set the Pseudo Automatic Number Identification (PANI) prompt to YES to display PANI.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ANI	Change Automatic Number Identification options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
....	....	
PANI	 YES (NO)	 Display. Do not display Pseudo Automatic Number Identification (default). ***Note:*** When PANI is set to NO (Do not display PANI), the PANI will display briefly, then disappear.
....	....	

LD 16 – Configure the M911 ANI format.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15

ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
DES	x...x	Designator field for trunk (0-16 character alphanumeric).
TKTP	DID	Direct Inward Dialing trunk data block.
M911_ANI	YES	Receive ANI digits for M911 route.
M911_TRK_TYPE	 (911T) 911E	Meridian 911 ANI trunk type. E911 tandem connections (default). End office connections.
M911_FORM	 2 (1)	M911 ANI format. II (2 digits) +10/20-digit ANI. NPD (1 digit) +7-digit ANI (default).
....	....	

Feature operation

No specific operating procedures are required to use this feature.

16-Button Digitone/Multifrequency Operation

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Feature description

This feature allows the use of a 2500-type telephone with 16 buttons instead of 12 buttons. The extra keys provide single button access to features that would otherwise require the use of Flexible Feature Codes. The feature also provides an autodial function. With this feature, autodial is also available to 12-button Digitone/Multifrequency (DTMF) telephones equipped with a true ground (GRD) button and 2500-type telephones with switchhook flash and calibrated flash.

Not all telephones must share the same assignments. In LD 18, functions can be overlay programmed against a key for each of the three modes. A set of these key-function definitions can then be assigned to one or more telephone station groups. Up to 127 sets of key function assignments (called ABCD tables) are permitted.

The following Flexible Feature Code functions can be accessed using the new (A, B, C, D, * and #) keys while in the pre-dial mode (when the telephone is receiving dial tone):

- authorization code
- automatic set relocation
- automatic wake-up activate
- automatic wake-up deactivate
- automatic wake-up verify
- Call Detail Recording charge account
- call forward all calls activate
- call forward all calls deactivate
- call forward all calls verify
- call forward toggle
- call park access
- conference diagnostics
- deactivate RGA, LND, SNR, or CFW
- electronic lock phone
- electronic lock phone (remote)
- Group Hunting pilot DN
- Incoming Call Identification (ICI) activate
- ICI deactivate
- ICI print
- integrated message system access
- last number redial
- maintenance access
- pick up DN
- pick up group

- pick up ringing number
- radio paging initiate (parallel)
- radio paging initiate (serial)
- radio paging answer (parallel)
- ring again deactivate
- ring again verify
- room status
- speed call controller
- speed call erase
- speed call user
- store number (erase)
- store number (redial)
- store number (save)
- system speed call user
- trunk answer from any station
- terminal diagnostics
- trunk verification, and
- user status.

The following functions can be accessed using the new (A, B, C, D, * and #) keys while in the post-dial mode (when it receives special dial tone after a recall during an active call, or after a busy DN has been dialed):

- Call Detail Recording charge account
- call park
- Conference six trunk disconnect
- ICI override
- last number redial
- Malicious Call Trace

- override
- permanent hold
- radio paging initiate (parallel)
- radio paging initiate (serial)
- ring again activation
- speed call user
- store number (redial)
- store number (save), and
- system speed call user.

Operating parameters

All Digitone Receivers (DTRs) on the system must have the correct strap settings for full 16-button DTMF detection.

An ABCD table must be defined, and associated with a station group.

The customer must have the SPRE code defined, in order to activate FFC functions through the A, B, C, and D keys.

The Multi-party Operations feature must be present if control digits are to be used.

The user needs a 16-button DTMF 2500-type telephone to make full use of this feature.

The 2500-type telephone must be defined as a member of a station group with an associated ABCD table.

All the requirements for the existing system, customer and station combination must be met.

Feature interactions

China – Flexible Feature Codes - Busy Number Redial

BNR allowed can be a postdial function, and BNR denied can be a predial function. Both FFCs may be dialed normally from a 16-button DTMF telephone.

China – Flexible Feature Codes - Customer Call Forward

CCFA and CCFD are allowed as predial ABCD functions. They can also be dialed normally from 16-Button DTMF telephones.

China – Flexible Feature Codes - Outgoing Call Barring

The Outgoing Call Barring FFCs are not allowed as ABCD functions. They can be dialed normally from 16-Button DTMF telephones.

Flexible Feature Codes

The Flexible Feature Codes (FFC) package must be installed, or the FFC functions are not available. However, control functions are still available. An FFC table must be defined for the customer, or the FFC functions are not available.

Group Hunt

Group Hunt Pilot DN (GRHP) function is not supported. Group Hunting and Speed Call DN Access can be accessed through the Autodial function.

Italian Central Office Special Services

The special service FFC is not supported on the ABCD keys of 16-button DTMF sets.

Feature packaging

16-Button Digitone/Multifrequency Telephone (ABCD) package 144.

Dependency:

- Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Modify the system hardware and software parameters to enable or disable the 16-Button Digitone/Multifrequency Operation feature.

2 LD 18 – Create or modify data for this feature in the 16-Button DTMF Data block.

LD 17 – Modify the system hardware and software parameters to enable or disable the 16-Button Digitone/Multifrequency Operation feature.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	PARM	System Parameters.
...		
PARM	(NO) YES	(No) Change to system parameters.
- ABCD	(NO) YES	16-Button DTMF (is not) is enabled.

LD 18 – Create or modify data for this feature in the 16-Button DTMF Data block.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	ABCD	16-Button DTMF data.
...		

Feature operation

Each button (A, B, C, D, * and #) can have up to three functions assigned to it. The function accessed when a key is pressed is determined by the mode of operation (pre-dial, post-dial or control mode). Functions are assigned to keys by way of overlay programs. The functions can be either Flexible Feature Code functions or the autodial function. An autodial number (of up to 23 digits) can be assigned to any of these buttons for either the pre-dial or post-dial modes. In addition, an autodial number can be assigned to the recall (RCAL) button in the pre-dial mode.

2 Mbps Digital Trunk Interface Enhancements

Contents

This section contains information on the following topics:

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Operating parameters 89
Feature interactions 89
Feature packaging 89
Feature implementation 90
Feature operation 90

Feature description

The following enhancements have been added to the existing 2 Mbps Digital Trunk Interface (DTI2) in order to meet various customer requirements.

Alarm Handling on Direct Inward Dialing Channels

If an alarm condition occurs on a Direct Inward Dialing (DID) channel, this enhancement delays the sending of connect and disconnect signals, until the alarm condition is cleared.

Alarm Handling on Incoming Public Exchange/Central Office or Direct Inward Dialing Trunks

This enhancement clears non-established calls on incoming Public Exchange/ Central Office (CO) or Direct Inward Dialing (DID) trunks when an alarm condition occurs. When the alarm condition is cleared, the calls are diverted to the attendant.

Call Clearance

This enhancement affects the handling of incoming and outgoing call clearance for Central Office (CO) calls.

Call Clearance is handled differently if the Clear Forward signal (CLRF) is defined, or if the Clear Forward signal and the IDLE signal do not have the same definition. The Call Clearance is also handled differently for outgoing and incoming calls.

For outgoing calls being disconnected by the system, a clear forward and then an IDLE signal is sent by the system. The call state determines when the IDLE signal is sent. If the call is answered, the IDLE signal is sent within 300 milliseconds of the reception of a clear back signal from the CO. If the outgoing call is not answered, the IDLE signal is sent after 800 milliseconds (plus or minus 50 milliseconds) of the clear forward signal being sent. If the CO answers during this 800 milliseconds period, the system continues to send the clear forward signal until it receives a clear back signal from the CO.

For outgoing calls being disconnected by the CO, a clear back signal is sent by the CO when it wishes to disconnect. The system then sends a clear forward signal within 300 milliseconds of having received the clear back signal, followed by an IDLE signal within 800 milliseconds (plus or minus 50 milliseconds) of having sent the clear forward signal.

For incoming calls being disconnected by the system, a clear back signal is sent by the system. Upon receiving a clear forward signal from the CO, the system sends an IDLE signal within 300 milliseconds of having received the clear forward signal.

For incoming calls being disconnected by the CO, a clear forward signal is sent by the CO when it wishes to disconnect. If the call is answered, the system sends a clear back signal within 300 milliseconds of having received the clear back signal from the CO, and then an IDLE signal after 800 milliseconds (plus or minus 50 milliseconds) of having sent the clear forward signal. If the call is not answered, the system sends an IDLE signal within 300 milliseconds of having received the clear forward signal from the CO.

If an alarm condition occurs while a clear forward or clear back signal is being sent for the 800 milliseconds time period, the system continues to send the signal until the alarm condition clears.

Clock Synchronization

This enhancement affects the clock synchronization controller. If a DTI loop enters its most severe alarm state (the No-New-Calls state), the system disables the clock port.

Direct Inward Dialing Call Offering

The Central Office (CO) operator will be able to offer a Direct Inward Dialing (DID) call to the attendant. When a DID call terminates on a busy station, and the End of Selection Busy (EOSB) signal has been sent to the CO by the analog (500/2500 type) telephone, the CO can then send an Operator Pulse Signal (OPRS) back to the analog (500/2500 type) telephone. This OPRS causes the analog (500/2500 type) telephone to forward the call on to the attendant.

Disable Out-of-Service Alarm State

This enhancement allows the system to disable the Out-of-Service (OOS) alarm state for an error, leaving the No New Call alarm state as the most severe state. This is done by setting the OOS threshold time for an error to zero.

Fault Signal

On an incoming call, if a Fault (FALT) Signal is received by the circuit switched network while in an IDLE state, the circuit switched network will respond with a Fault Signal until the CO returns to the IDLE state. On an outgoing call, the circuit switched network will enter the FALT state if a Release Control (RCTL) signal is not received within 30 seconds.

Incoming Seizure

This enhancement, applied on a group basis, allows the Central Office to initiate a call from a lockout or far-end fault state.

Outpulsing Delay

This enhancement provides a delay before outpulsing on 2 Mbps DTI trunks.

Release Control

The circuit switched network will now be able to send and receive the Release Control (RCTL) signal, which is sent by the called party on both incoming and outgoing calls to indicate disconnection is complete. The RCTL signal is sent by either the CO or circuit switched network in response to a Release Clear Forward signal.

Signal Recognition

This enhancement gives the system more flexibility in handling receive signals. The system can recognize a signal based on the ABCD signaling bits. Any non-significant signaling bits of a receive signal can be flagged as do-not-care. The system can then ignore these do-not-care bits before trying to determine which signal it has received.

64 Kbit Alarm Indication Signal Handling

This enhancement adds the 64 Kbit Alarm Indication Signal (AIS) as a sixth group II error state. This error state is treated the same as the other group II error states.

Centre National d'Études des Télécommunications enhancement for trunks entering an alarm state

This enhancement requires the QPC915 and ensures compliance with the Centre National d'Études des Télécommunications (CNET) requirements for trunks entering an alarm state.

Trunks entering an alarm state are processed according to the type of trunk they are configured as and their previous state.

For all cases, signaling will not occur on the trunk while it is in an alarm state.

Idle trunk

When an idle trunk enters an alarm state, it will not send the "FAULT" signal.

DID trunk

Trunk seized and receiving digits

The call is taken down and the trunk is idled.

Call initiated but not answered

A timer is started when the alarm state is entered, its duration is between 20 and 40 seconds, and the called set continues to ring. During this time one of three cases may occur:

- **The timer expires:** the call is disconnected, all resources but the incoming trunk are released (delayed disconnect). This occurs even if the following case has already happened.
- **The called set answers:** no affect on the timer; the delayed disconnect will occur if the alarm is not cleared.
- **The alarm stops:** no affect on the connection, the timer is stopped and reset, and delayed signals are sent to the far end.

Call answered

The call is not dropped upon entering an alarm state. If the near-end party goes on-hook during alarm, the party is released and all resources are idled except the trunk, which is put in a delayed disconnect state.

Disconnect

The alarm is ignored with respect to internal system processing, and the trunk is put in a delayed disconnect state.

Outgoing Central Office Trunk (COT) call

If the destination has not answered, no action is taken when entering an alarm state. If the originator goes on-hook during an alarm state, the disconnect signal is delayed.

If the destination goes on-hook while in an alarm state, the software waits for the originator to go on-hook also. If the alarm is still present when the originator goes on-hook, system resources are idled, but the trunk is left in a delayed disconnect state.

Incoming COT call

Call initiated

When entering an alarm state, the call is disconnected and all system resources are idled, including the trunk itself.

If the Attendant or Night set answered before the trunk entered the alarm state, the call is connected and the “CONNECT” signal is delayed until the alarm state is cleared.

Disconnect

The system completes the disconnect and idles the trunk without waiting for an “IDLE” signal from the far end.

Centre National d’Études des Télécommunications enhancement for trunk cards exiting an alarm state

This enhancement requires the QPC915 and ensures compliance with the Centre National d’Études des Télécommunications (CNET) requirements for cards exiting an alarm state.

At the end of a group I alarm state, the software requires the pack to send the ABCD status of each configured trunk. At the end of a group II alarm state, the software receives a report of valid ABCD status after having received a confirmation from the firmware that the firmware is functioning as expected. The system software state is updated according to this report.

Processing of overload conditions

Several enhancements occur:

- When receiving more than 100 messages per second from a 2 Mbps Digital Trunk Interface (QPC915) pack, the system attempts to go into No New Call (NNC) state and disables the error reporting. A DTA320 message is printed on the Maintenance Terminal to inform the technician. After at least two seconds have elapsed, the error reporting is re-enabled and a DTA321 message is printed. If this situation repeats itself more than 20 times within the next two minutes, the pack is disabled.
- The software status is updated to reflect the firmware status after overload.
- The overload process is able to recognize the channel causing the overload when the case arises.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There are no specific packaging requirements associated with this feature.

Feature implementation

Note: LD 73 is modified to allow the implementation of the CNET enhancement for trunks entering an alarm state and trunk cards exiting an alarm state. The enhancement is implemented by responding YES to the new FRFW prompt in LD 73.

LD 73 – Implement the system hardware and software parameters.

Prompt	Response	Description
REQ	NEW CHG	Add, or change
TYPE	DTI2	2.0 Mbps DTI.
GP2	T2 mt dt ct ot	Group 2 error thresholds.
FRFW	(NO) YES	French Firmware. Enter YES to enable the CNET enhancement for trunks entering an alarm state processing capabilities. Requires that QPC915 packs be equipped. Enter YES to enable the CNET enhancement for trunk cards exiting an alarm state processing. Requires that QPC915 packs be equipped. Enter NO if the CNET enhancement for trunks entering an alarm state processing capabilities are not required. Enter NO if the CNET enhancement for trunk cards exiting an alarm state processing is not required. Default is NO.

Feature operation

No specific operating procedures are required to use this feature.

2 Mbps Digital Trunk Interface

Contents

This section contains information on the following topics:

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Feature interactions 92
Feature packaging 92
Feature implementation 92
Feature operation 94

Feature description

The 2 Mbps Digital Trunk Interface (DTI2) feature provides digital connectivity between a system digital network loop and an external digital carrier termination. It provides digital speech on up to 30 channels at 2 Mbps on one system loop and the bipolar carrier terminal. Within the system, the DTI2 operates as a general purpose sender and receiver of ABCD (signaling) bits. The DTI software sets the ABCD bits to represent the appropriate signaling for the trunk being supported.

For 2 Mbps DTI, use the QPC775 clock controller.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Periodic Pulse Metering

Periodic Pulse Metering operates the same for 2 Mbps DTI as for analog trunks.

Pulsed E&M DTI2 Signaling

Pulsed E&M DTI2 signaling is based on 2 Mbps DTI.

Feature packaging

2 Mbps Digital Trunk Interface (DTI2) package 129.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Create or modify trunk data blocks for DTI2 on a per trunk basis.

2 LD 16 – Create or modify DTI2 trunk route data blocks.

3 LD 17 – Modify the system hardware and software parameters to enable or disable the feature.

4 LD 73 – Implement the system hardware and software parameters to enable or disable the DTI feature.

LD 14 – Create or modify trunk data blocks for DTI2 on a per trunk basis.

Prompt	Response	Description
REQ	NEW CHG	Add, or change
TYPE	a...a	Type of data block.

SICA	(1)-16	Signaling Category table number. The category must already be defined in LD 73. Default is 16 if loop type = Japanese Digital Multiplex Interface (JDMI).
PDCA	(1)-16	Pad Category table number. The PAD category must already be defined in LD 73. Default is 16 if loop type = JDMI.
PCML	MU A	Indicate whether Mu-law or A-law Pulse Code Modulation (PCM) for voice calls is active in the channel. Not prompted for JDMI loops.

LD 16 – Create or modify DTI2 trunk route data blocks.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	a...a	Route type.
DTRK	(NO) YES	Digital trunk route.
DGTP		Digital trunk type.
	(DTI) PRI DTI2 PRI2 JDMI	1.5 Mbps DTI (default). 1.5 Mbps Primary Rate Interface. 2 Mbps DTI. 2 Mbps Primary Rate Interface. Japanese Digital Multiplex Interface. Prompted when the DTI2 or PRI2 package is equipped.

LD 17 – Modify the system hardware and software parameters to enable or disable the feature.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CEQU	Common Equipment Parameters.

...		
- DTI2	0-159	2 Mbps Digital Trunk Interface (DTI) loop number. Prompted the when DTI2 or PRI2 package is equipped.

LD 73 – Implement the system hardware and software parameters to enable or disable the DTI feature.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DTI2	2 Mbps DTI.
...		

Feature operation

No specific operating procedures are required to use this feature.

2500 Telephone Features

Contents

This section contains information on the following topics:

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Feature implementation 96
Feature operation 97

Feature description

This feature allows 2500 telephones (that is, basic push-button sets having no feature keys) to access features otherwise available only on Meridian 1 proprietary telephones. By dialing an octothorpe (#) and a single-digit access code, 2500 telephones can access the following features:

- Call Forward All Calls Dial #1
- Speed Call Controller Dial #2
- Speed Call User Dial #3
- Permanent Hold Dial #4

Operating parameters

Allow or deny the Call Forward All Calls, Speed Call Controller, Speed Call User, and Permanent Hold features in LD10.

Except for the access codes used, feature operation is the same as for Meridian 1 proprietary telephones.

Feature interactions

500 Telephone Features

When 500 Set Dial Access to Features (SS5) package 73 is equipped, 2500-type telephones also access features by dialing SPRE and a two-digit access code as follows:

- System Speed Call User SPRE + 73
- Call Forward All Calls SPRE + 74
- Speed Call Controller SPRE + 75
- Speed Call User SPRE + 76
- Permanent Hold SPRE + 77

Remote Call Forward

When Flexible Feature Codes (FFC) package 139 is defined and active on your system, a telephone provisioned for Call Forward in LD 10 can also Call Forward All Calls from a remote internal DN.

Feature packaging

There are no specific packaging requirements associated with this feature.

Feature implementation

LD 10 – Enable 2500 Telephone Features.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA	(Deny) allow transfer.
FTR	CFW xx	Call Forward All Calls and DN length (4-23). Enter X CFW to remove.
	SCC xxxx	Speed Call Controller and list number. Enter X SCC to remove.
	SCU xxxx	Speed Call User and list number. Enter X SCU to remove.
	SSU xxxx	System Speed Call User and list number. Enter X SSU to remove.
	PHD	Allow Permanent Hold. Enter X PHD to remove.

Feature operation

Call Forward All Calls

Case 1: FFC active, CFW not active

On a telephone with Flexible Feature Codes implemented, but without Call Forward currently active, use these steps to activate the feature:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Dial the DN where you want calls to be forwarded. The dial tone disappears.
3. Hang up to complete the activation.

To deactivate Call Forward, follow these steps:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Hang up to complete deactivation.

Case 2: FFC not active, CFW not active

On a telephone without Flexible Feature Codes or Call Forward currently Active, use these steps to activate the feature:

1. Lift the handset and dial #1. You hear dial tone.
2. Dial the DN where you want calls to be forwarded. The dial tone disappears.
3. Hang up to complete the activation.

To deactivate Call Forward, follow these steps:

1. Lift the handset and dial #1. You hear dial tone.
2. Hang up to complete deactivation.

Case 3: FFC active, CFW active

On a telephone with Flexible Feature Codes and Call Forward currently active, use these steps to deactivate the feature:

1. Lift the handset and dial #1. You hear confirmation tone.
2. Hang up to complete the deactivation.

To reactivate Call Forward, follow these steps:

1. Lift the handset and dial #1. You hear dial tone.
2. Dial the DN where you want calls to be forwarded. The dial tone disappears.
3. Hang up to complete the activation.

– or –

1. Lift the handset and dial #1. You hear dial tone.
2. Dial the DN where you want calls to be forwarded. The dial tone disappears.
3. Dial the EOD string. You hear a confirmation tone.
4. Hang up to complete the activation.

– or –

1. Lift the handset and dial #1. You hear dial tone.
2. Hang up to complete the activation. Calls are forwarded to the last Call Forward DN used by this telephone.

Speed Call Controller

To update a predefined Speed Call list, follow these steps:

1. Lift the handset and dial #2. You hear dial tone.
2. Dial the Speed Call code (0-999), followed by the telephone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.
3. Hang up.

To change a number associated with a list, follow these steps:

1. Lift the handset and dial #2. You hear dial tone.
2. Dial the Speed Call code (0-999), followed by the new telephone number. The new number automatically replaces the old one. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.
3. Hang up.

To remove an entry from a Speed Call list, follow these steps:

1. Lift the handset and dial #2. You hear dial tone.
2. Dial the Speed Call code (0-999) you want to remove.
3. Hang up.

Speed Call User

To make a Speed Call, follow these steps:

1. Lift the handset and dial #3. You hear dial tone.
2. Dial the Speed Call code (0-999).
3. The number is dialed automatically.

System Speed Call User

To make a System Speed Call, follow these steps:

1 Lift the handset and dial SPRE 73. You hear dial tone.

2 Dial the System Speed Call code (0-999).

3 The number is dialed automatically.

Permanent Hold

To activate Permanent Hold while on a call, follow these steps:

1 Flash the switchhook. You hear dial tone.

2 Dial #4.

3 Hang up.

The call remains on hold until you lift the handset again or the other party disconnects.

500 Telephone Features

Contents

This section contains information on the following topics:

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Feature packaging 102
Feature implementation 102
Feature operation 103

Feature description

This feature allows 500-type (rotary dial) telephones to use Call Forward, Speed Call, and Permanent Hold. Since 500-type telephones do not have an octothorpe (#), the following features are activated by dialing SPRE and a two-digit access code.

- System Speed Call SPRE + 73
- Call Forward All Calls SPRE + 74
- Speed Call Controller SPRE + 75
- Speed Call User SPRE + 76
- Permanent Hold SPRE + 77

Operating parameters

Allow or deny the System Speed Call, Call Forward All Calls, Speed Call Controller, Speed Call user, and permanent hold features in LD 10.

Except for the SPRE codes used, feature operation is the same as for Meridian 1 proprietary telephones.

Feature interactions

2500 telephone features

When Special Service for 2500 Sets (SS25) package 18 is equipped, 2500 telephones also access the above listed features by dialing the SPRE and a two-digit access code.

Feature packaging

500 Set Dial Access to Features (SS5) package 73 requires Special Service for 2500 Sets (SS25) package 18.

Feature implementation

LD 10 – Enable 500 type telephone features.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA	(Deny) allow transfer.

FTR	CFW xx	Call Forward All Calls and DN length (4-23). Enter X CFW to remove.
	SCC xxxx	Speed Call Controller and list number. Enter X SCC to remove.
	SCU xxxx	Speed Call User and list number. Enter X SCU to remove.
	SSU xxxx	System Speed Call User and list number. Enter X SSU to remove.
	PHD	Allow Permanent Hold. Enter X PHD to remove.

Feature operation

Call Forward All Calls

To forward your calls, follow these steps:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Dial the DN to where you want your calls forwarded.
3. Hang up.

To cancel forwarding, follow these steps:

1. Lift the handset and dial SPRE + 74. You hear dial tone.
2. Hang up.

Speed Call Controller

To update a predefined Speed Call list, follow these steps:

1. Lift the handset and dial SPRE + 75. You hear dial tone.
2. Dial the Speed Call code (0-999), followed by the telephone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear fast busy tone.
3. Hang up.

To change a number associated with a list, follow these steps:

1. Lift the handset and dial SPRE + 75. You hear dial tone.

2 Dial the Speed Call code (0-999), followed by the new telephone number. The new number automatically replaces the old one. If the entry is accepted, you hear silence. If the entry is not accepted, you hear fast busy tone.

3 Hang up.

To remove an entry in a Speed Call list, follow these steps:

1 Lift the handset and dial SPRE + 75. You hear dial tone.

2 Dial the Speed Call code (0-999) you want to remove.

3 Hang up.

Speed Call User

To make a Speed Call, follow these steps:

1 Lift the handset and dial SPRE + 76. You hear dial tone.

2 Dial the Speed Call code (0-999).

3 The number is dialed automatically.

System Speed Call User

To make a System Speed Call, follow these steps:

1 Lift the handset and dial SPRE + 73. You hear dial tone.

2 Dial the System Speed Call code (0-999).

3 The number is dialed automatically.

Permanent Hold

To activate Permanent Hold while active on a call, follow these steps:

1 Flash the switchhook. You hear dial tone.

2 Dial SPRE + 77.

3 Hang up.

The call remains on hold until you lift the handset again or the other party disconnects.

500/2500 Line Disconnect

Contents

This section contains information on the following topics:

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Feature interactions 110
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Feature implementation 112
Feature operation 113

Feature description

500/2500 Line Disconnect

500/2500 Line Disconnect is invoked when the system detects on-hook/ disconnect supervision from a party connected to an analog (500/2500-type) port. Dial tone is sent to this port for a specified period of time (the default is six seconds) which is defined in LD 15 at the Line Disconnect Tone Timer (LDTT) prompt.

It is used when the analog (500/2500-type) port is connected to an automated attendant or voice mail. It allows the system to know that it is not connected to a telephone, and to disconnect if the other telephone has hung up (for example, during an automated message or a voice mail message).

An analog (500/2500-type) port with LDTA Class of Service receives disconnect tone in the following cases:

- an incoming internal call is placed to an LDTA port and then disconnects
- incoming call from a trunk with disconnect supervision is placed to an LDTA port and then the incoming trunk disconnects, or
- an internal DN places an outgoing call on a trunk with disconnect supervision, then transfers the call to the LDTA port and then the trunk disconnects.

Figure 1 illustrates how an incoming trunk call or internal call functions with 500/2500 Type Line Disconnect. This illustration shows the incoming trunk call or internal call disconnected and dial tone being provided by the analog (500/2500-type) port with the new Class of Service (CLS) Line Disconnect Tone Allowed (LDTA).

Figure 1
Incoming trunk call of internal call disconnect function when 500/2500 line disconnect feature is configured

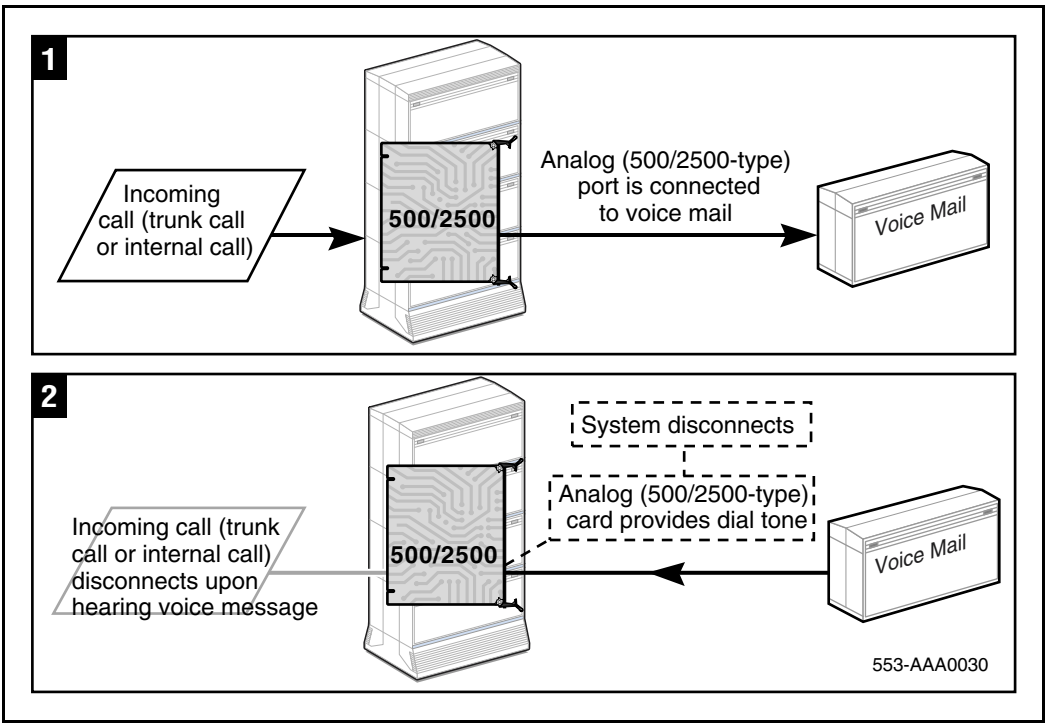
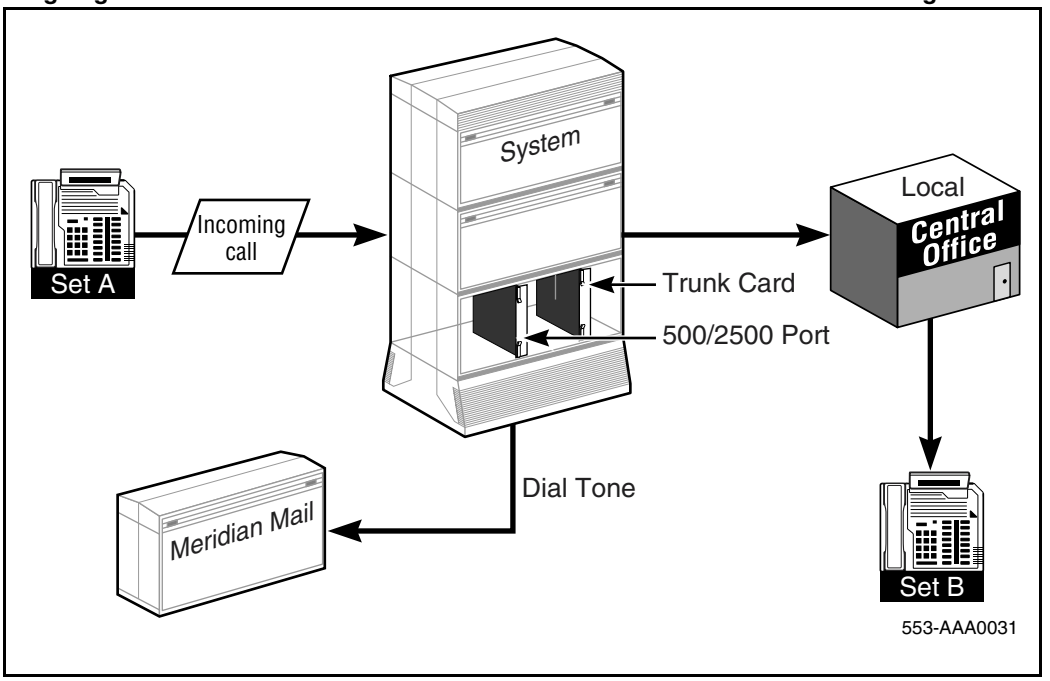


Figure 2 illustrates how an outgoing call functions with this feature. This illustration shows an outgoing call from the system to the Central Office. Station A transfers Station B to Meridian Mail and goes on-hook. When Station B disconnects, dial tone is provided by the analog (500/2500-type) port with the new LDTA Class of Service.

Figure 2
Outgoing call disconnect function when 500/2500 line disconnect feature is configured



500/2500 Line Disconnect for Outgoing Calls

When devices such as dictation machines are connected to an analog (500/2500-type) line port, they rely on detecting a tone to indicate that the far end has released. This is necessary because the line conditions on an analog (500/2500-type) circuit do not change regardless of the status of the far end.

Currently, when a system detects an on-hook/disconnect supervision signal from a party on a trunk that provides disconnect supervision, and the trunk is connected to an analog (500/2500-type) port with the Line Disconnect Tone Allowed (LDTA) Class of Service, dial tone is sent for the time specified in the Customer Data Block. Thus, the device physically connected to the 500/2500 port disconnects itself and the line port as well. This functionality is used in applications requiring predictive dialing; however, previously it was limited to incoming calls.

The 500/2500 Line Disconnect for Outgoing Calls feature expands the 500/2500 Disconnect capability to encompass outgoing calls.

Operating parameters

500/2500 Line Disconnect

Line Disconnect Tone is not provided on outgoing calls from the LDTA port.

500/2500 Line Disconnect for Outgoing Calls

This feature only works with internal calls or with trunks that provide disconnect supervision. If a trunk is used that does not have disconnect supervision, the system does not detect the far end disconnection and the release of the call is still dependent upon the internal timing of the Automated Dialing Equipment.

This feature only applies to Automated Dialing Equipment systems capable of recognizing dial tone as a disconnect signal.

When an analog (500/2500-type) port is receiving a disconnect dial tone, it is not possible to dial a number. Dial tone cannot be broken. The port has to be released before dialing out.

Feature interactions

500/2500 Line Disconnect

Attendant Extended Call

500/2500 Line Disconnect applies if the attendant extends a call to an analog (500/2500-type) port that is connected to a Voice Response Unit (VRU); or the attendant extended a call to an analog (500/2500-type) port that is connected to a VRU and remains in the call, and the other party has disconnected.

Conference
No Hold Conference

If one of the parties in the conference is connected to an analog (500/2500-type) port that is in turn connected to a VRU, dial tone is provided to the analog (500/2500-type) port when all the other parties in the conference disconnect. This feature enhancement applies in the same way to Call Transfer and Hunting.

500/2500 Automatic Call Distribution agent

If a call is involved with an analog (500/2500-type) Automatic Call Distribution (ACD) agent that is connected to a VRU and the other party has disconnected, 500/2500 Line Disconnect applies. When the other party disconnects, the analog (500/2500-type) agent will be returned to the idle agent queue.

500/2500 Line Disconnect for Outgoing Calls

Attendant Extended Call

The 500/2500 Line Disconnect for Outgoing Calls feature applies if an attendant extends a call originated from an analog (500/2500-type) line port with LDTA Class of Service to a trunk or an internal extension, and the attendant has disconnected from the call. When the far end disconnects and this is a simple call, dial tone is provided to the analog (500/2500-type) line port.

Call Forward All Calls
Call Forward No Answer
Call Forward Busy
Call Forward by Call Type

The 500/2500 Line Disconnect for Outgoing Calls feature applies if a call originated from an analog (500/2500-type) line port with LDTA Class of Service is Call Forwarded to a trunk or another internal extension.

Call Transfer

The 500/2500 Line Disconnect for Outgoing Calls feature applies if a call originating from an analog (500/2500-type) line port with LDTA Class of Service is transferred by the called party to a trunk or another internal extension.

Conference
No Hold Conference

If Automated Dialing Equipment is connected to an internal extension that uses transfer or conference to include a trunk or another internal extension in the call, dial tone will be provided to the analog (500/2500-type) port when all the other parties disconnect.

Hunting

The 500/2500 Line Disconnect for Outgoing Calls feature applies if a call originated from an analog (500/2500-type) line port with LDTA Class of Service reaches a busy telephone that hunts to a trunk or to another internal extension.

Tone to Last Party

With the Tone to Last Party (TLP) feature configured, tones given to telephones, whether involved in an internal or external call, are defined in the Tone Tables defined for the customer. If the TLP timer in the tone table is set to zero, the feature is disabled. If the TLP timer has a value greater than zero, this feature is active for all analog (500/2500-type) telephones at the customer location. The 500/2500 Line Disconnect feature takes precedence if the Tone to Last Party feature is enabled for a customer and the analog (500/2500-type) telephone has LDTA Class of Service.

Analog (500/2500-type) Automatic Call Distribution Agents

If an Automated Dialing Equipment (ADE)/Voice Response Unit (VRU) is involved in a call with an analog (500/2500-type) Automatic Call Distribution Agents (ACD) agent and the party disconnects, the ADE will be provided dial tone when the last party (except for the ADE/VRU) has disconnected.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Allow Line Disconnect Tone for analog (500/2500-type) ports.

2 LD 15 – Specify the dial tone timer for analog (500/2500-type) ports.

Note: Feature implementation is the same for both 500/2500 Line Disconnect and 500/2500 Line Disconnect for Outgoing Calls.

LD 10 – Allow Line Disconnect Tone for analog (500/2500-type) ports.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(LDTD) LDTA (WTA) WTD	(Deny) allow Line Disconnect Tone. (Allow) deny Warning Tone.

LD 15 – Specify the dial tone timer for analog (500/2500-type) ports.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	TIM	Timers.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- LDTT	2-(6)-30	Line Disconnect Tone timer for the 500/2500 port, in seconds.

Feature operation

No specific operating procedures are required to use this feature.

AC15 Recall: Timed Reminder Recall

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Feature description

The AC15 Timed Reminder Recall feature allows timed recall functionality in an environment where a system is used as a hub for systems that are connected with an AC15 TIE trunk.

The feature enables a call established with a local system set or trunk and extended by a controlling party over an AC15 TIE trunk to be recalled after a programmed period of time to the system attendant. The controlling party is an attendant or a set connected to the system. When Night Service is activated, the call will be recalled to the Night DN if the original call is external and the International Supplementary Features (SUPP) package 131 is equipped.

Operating parameters

The call must be extended to the AC15 TIE trunk by a controlling party on the system. The feature is not applicable to tandem calls through the system, calls routed directly by system routing controls, and direct calls over the AC15 TIE trunk.

AC15 TIE trunks must be configured on a route basis.

Night Service must be activated, the original call must be an external call, and International Supplementary Features (SUPP) package 131 equipped for a call to be recalled to the Night DN. That is the only situation where an AC15 recall will not be presented to the attendant.

Answer supervision must be configured on the AC15 TIE trunk for the feature to be activated.

XFEM trunk cards that support AC15 signaling are required (for example, the NT5K19AC trunk card for the UK).

Feature interactions

AC15 Recall: Transfer from Norstar

A transfer performed by an AC15 trunk using the Transfer from Norstar feature to another AC15 trunk is not subject to Timed Reminder Recall. This is to prevent a call transferred by someone on Norstar from recalling the system attendant.

It is recommended that all AC15 cards on the network's system are NT5K19AC or later. This is mandatory for the system which directly interfaces with the Norstar (this requirement applies to all of this switch's AC15 cards, even to those that do not directly interface with the Norstar).

Access Restrictions

With call modification, a trunk-to-trunk connection is controlled by signaling, recall capability and the supervision assigned to each trunk. For example, an established call from an unsupervised trunk cannot be transferred over another trunk.

When the AC15 Timed Reminder Recall feature is to be activated, an established call with an unsupervised trunk may be extended over an AC15 trunk because the connection is controlled before the called party answers by the AC15 recall timer.

Attendant Clearing During Night Service (ACNS)

If ACNS is active and there is a call being extended over an AC15 TIE trunk, when the attendant goes into Night Service, the transfer is completed and the feature is activated.

If there is an AC15 recall presented to the attendant and it goes in Night Service, the recall is put in the attendant queue.

If an AC15 recall has been answered by the attendant and it goes in Night Service, the call is removed from the attendant port and the feature is activated again.

Attendant Console

The Incoming Call Indicator (ICI) can be configured to work with this feature. When there is a recall, the ICI RLL key lamp is updated, and is either lit or flashing. The attendant can answer the recall by pressing the ICI RLL key instead of the Loop key.

Attendant Console – Call Key Lamp State and Display

When the attendant is dialing over an AC15 TIE trunk and the AC15 Timed Reminder Recall feature is to be activated, the destination lamp state is winking instead of lit. It is only lit when the called party answers.

Attendant Forward No Answer

If the Attendant Forward No Answer feature is activated and the attendant fails to answer, the attendant is forced into Busy Position and the call goes to the first idle attendant or is put into the attendant queue. If the conditions are also satisfied to put the customer in Night Service and the original call is an external call, the AC15 recall is directed to the Night DN.

Attendant Overflow Position

AC15 recalls are not routed to the Attendant Overflow Position. They are directed to the first idle attendant or put in the attendant queue.

Attendant Secrecy

Secrecy is not activated when AC15 recalls are presented to the attendant.

Call Hold, Permanent

Call Hold Permanent is activated when the attendant presses the HOLD key then the Release (RLS) key when extending a call, the call will then be permanently held on the Loop key. If the attendant retrieves the original call on hold by pressing the Loop key, the recall timer is stopped. If the attendant then presses the RLS key, the call is extended and the recall timer is restarted.

Called Party Name Display

When the AC15 recall is presented to an attendant or a set with a display, the source and destination names are shown beside the DNs or the ACODs.

Conference

The conference feature is sometimes used to perform a transfer when a controlling party establishes a call, the controlling party establishes a conference with a third party and releases, and a call is established between the two remaining parties.

If an established call is extended over a trunk to initiate a conference call, this conference call cannot be set up if this trunk has answer supervision and the called extension has not answered. The AC15 Timed Reminder Recall feature cannot be activated by using the conference feature to extend a call over an AC15 TIE trunk, because the AC15 TIE trunk must have answer supervision and the called extension must be ringing.

Network Attendant Service

If Night Service and Network Attendant Service are active, the recall is routed to a remote attendant. The original party is kept, the destination party is disconnected and the AC15 TIE trunk is released.

Night Service Enhancements

This feature is used to direct the call to the Night DN if the original call is an external call and the SUPP package 131 is equipped. When there is an AC15 recall and the attendant is in Night Service, the called party is disconnected (the AC15 trunk is released) and the original call is presented to the Night DN.

Periodic Clearing

When the Periodic Clearing feature is active, the Disconnect timer will interfere with the AC15 recall timer. The Disconnect timer is activated on a TIE trunk or an incoming Direct Inward Dialing (DID) or Central Office (CO) trunk which is connected to the AC15 TIE trunk. If the Disconnect timer expires first, the AC15 recall is cancelled and the trunk is disconnected. This is the case with a call which has been established with a TIE trunk or an incoming call on a DID or CO trunk that has been extended over an AC15 TIE trunk with the timed recall activated.

Recall to Same Attendant

With the AC15 Timed Reminder Recall feature, if Recall to Same Attendant = RSAA the call is presented to the attendant who last extended the call, if RTSA = RSAX the call is presented to the attendant who last extended the call or put in the queue if this attendant is busy.

Secrecy Enhancement

When the attendant answers an AC15 recall, the destination party is excluded from the connection. The attendant is connected to the source party and the excluded destination lamp is lit to show the exclusion of the destination party.

Series Calls

Series Calls cause a source call that has been extended to a local destination party to be recalled to the attendant when the destination party hangs up. In activating the AC15 Timed Reminder Recall, the called party is not local. Therefore, the Series Calls feature is not applicable.

Set Digit Display

When an AC15 recall is directed to the Night DN, if the Night DN set has a display, the display shows the external trunk and the AC15 trunk information.

Slow Answer Modification (SLAM)

With the AC15 Timed Reminder Recall feature, if SLAM is allowed, when the attendant answers an AC15 recall the destination party is disconnected and the AC15 TIE trunk is released.

Feature packaging

The AC15 Recall (ACRL) package 236 must be equipped to activate the AC15 Timed Reminder Recall feature.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Set the Slow Answer Recall timer at the RTIM prompt.

2 LD 16 – Define a TIE route and set the ATRR option.

3 LD 14 – Define an AC15 TIE trunk on an XFEM card

LD 15 – Set the Slow Answer Recall timer at the RTIM prompt.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	TIM	Timers Data Block.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- RTIM	xxx yyy zzz	xxx = timer in seconds for the Slow Answer Recall and the AC15 Timed Reminder Recall. yyy = timer in seconds for Camp-on Recall. zzz = timer in seconds for Call Waiting Recall.

LD 16 – Define a TIE route and set the ATRR option.

Prompt	Response	Description
REQ	NEW	New.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE	Trunk type.
...		
DTRK	NO	Digital trunk.
...		
TIDY	xxxx yyyy	Trunk identity.
ATRR	YES	AC15 Recall: Timed Reminder Recall. Calls transferred to an AC15 trunk on this route are subject to Timed Reminder Recall. Prompted with ACRL package 236 if TKTP = TIE and DRTK = NO.

LD 14 – Define an AC15 TIE trunk on an XFEM card

Prompt	Response	Description
REQ	NEW	New.
TYPE	TIE	TIE trunk.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	4D	Card density.
CUST	xx	Customer number, as defined in LD 15
...		
SIGL	WR4	AC15 signaling.
...		
SUPN	YES	Answer and disconnect supervision required.

Feature operation

Activate Timed Reminder Recall

The activation of the feature depends on whether the controlling party is the attendant or a set.

Attendant as a controlling party

1 A call is established on the source side of the attendant. The lamps displays appear as follows:

Loop is **lit**
Source is **lit**
Destination is **dark**
Rls key is **dark**

2 Dial an extension over an AC15 TIE trunk on the destination side. The lamps displays appear as follows:

Loop is **lit**
Source is **lit**
Destination is **winking**
Rls key is **dark**

3 Press the RLS key before the extension is answered. The AC15 recall timer is started. The lamps displays appear as follows:

Loop is **dark**
Source is **dark**
Destination is **dark**
Rls key is **lit**

Note: If the called extension answers the call, the recall timer is stopped.

Set as a controlling party

1 A call is established with a set on the system.

2 Transfer to an extension over an AC15 TIE trunk by using a flash hook on an analog (500/2500 type) telephone or pressing the TRN key on a proprietary telephone.

3 Complete the transfer before the extension answers by going on-hook on an analog (500/2500 type) telephone or pressing the TRN key on a proprietary telephone. This will start the AC15 recall timer.

Answer a Recall

Attendant

1 The recall rings the attendant. The original call is put on the source side and the destination party is put on the destination side. The lamps displays appear as follows:

Loop is **dark**
Source is **flashing**
Destination is **winking**
Rls key is **dark**

Note: If the called extension answers, the recall is removed from the Attendant Console.

2 Answer the recall. The called extension is still ringing on the destination side. The lamps displays appear as follows:

Loop is **lit**
Source is **lit**
Destination is **winking**
Rls key is **dark**

Pressing the Rls key at this point will reactivate the feature.

If the called extension answers the call after the attendant has picked up the recall, the originating party is kept on the source side and the destination party on the destination side of the attendant. A conference will occur between the attendant, the source, and the destination party. If the attendant releases, a normal call will then be established.

Night DN or Central Answering Position (Small System)

A Central Answering Position (CAP) is used as an alternative to an attendant on a system, particularly a Small System, which is not equipped with an Attendant Console. Any customer appears in Night Service and the CAP DN is the Night DN in this configuration. For the Night DN or the CAP operation, the following applies:

- For the original call to be directed to the Night DN, the call must be a direct CO/DID call or a DID/CO call through a Digital Private Network Signaling System (DPNSS1) or Network Attendant Service (NAS) ISDN trunk.
- For recall to the Night DN, the destination party is disconnected before the recall is presented to the Night DN.

AC15 Recall: Transfer from Meridian 1

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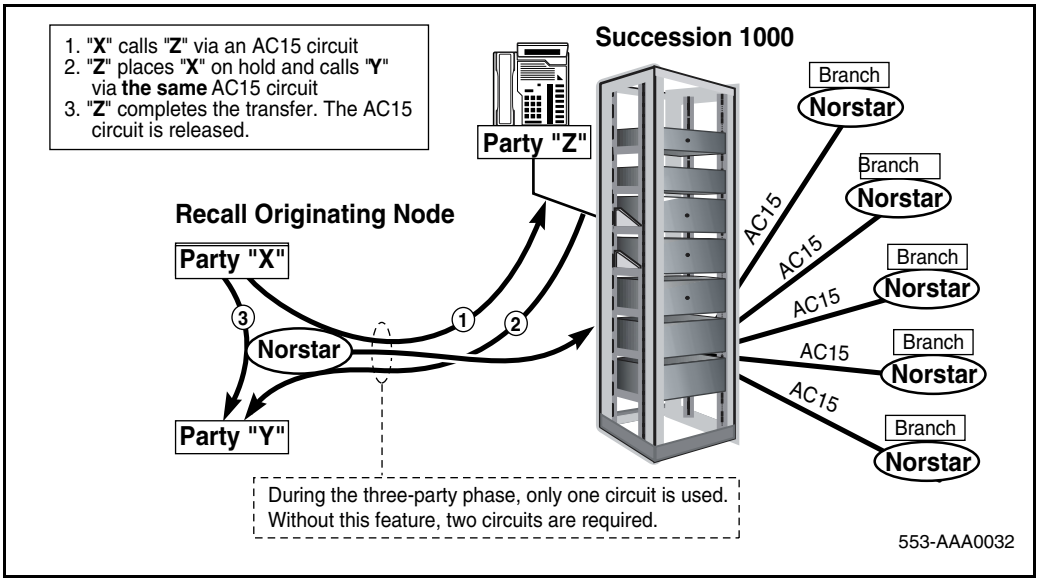
Feature description

The AC15 Recall: Transfer from Meridian 1 (ACRL) allows the Succession 1000M, Succession 1000, or Meridian 1 system to function as a "recall originating node" in situations where the Norstar functions as a control node. This capability permits signaling over AC15 trunks, which minimizes the number of AC15 circuits, optimizes the use of AC15 TIE trunks and avoids tromboning connections.

When a call with a party on the Norstar is transferred from the system, the ACRL feature enables the system to send a recall signal to the Norstar. This recall message permits the re-use of the same AC15 circuit, on which the call was received, to transfer the call. When a transfer is completed, the AC15 trunk is released. The following scenario demonstrates the ACRL feature capabilities.

A call occurs between party X (external call) and party Z (on the system). Party Z initiates the transfer feature and a recall signal is sent over the AC15 trunk. This signal is detected by the control node which puts the calling party X on hold and provides a dial tone to party Z to invoke a call transfer. The transfer dialed digits are sent on the AC15 trunk to the control node. A new call to party Y is placed that connects party Z with party Y. A release signal is sent when party Z completes the transfer and the AC15 trunk is released. Figure 3 illustrates this example.

Figure 3
System to Norstar call transfer



Operating parameters

AC15 Recall: Transfer from Meridian 1 requires XFEM trunk card NT5K19AC or later. This feature is only available in countries that use this card type.

There is no signaling capability for the control node to inform the tandem node or recall originating node that a party has answered or that there has been a release of any call on a split line.

The recall signal received on an AC15 trunk is not tandemed. No recall signal is sent on the reception of a “recall” in message.

Unsplitting of lines is not supported. In instances where a line is split, the line remains split until the whole trunk is released. This parameter ensures consistency on both sides of the AC15 channel. Additionally, it eliminates the possibility of selective release of a call in split mode.

The AC15 trunk must be configured with a digitone (CLS = DTN) Class of Service (LD 14) to ensure that the recall signal is received by the trunk. The trunk must also be configured on a modified XFEM trunk card.

The far end control node must be a switch that supports the recall signal, such as a Norstar.

Transfer chaining is not possible. There is no way to know if party X or party Y has gone on-hook once a trunk has been split. Therefore, Party Z cannot transfer to another set or initiate another consultation to a party on another node.

Electronic Switched Networks are supported on the initial transfer, provided that digits are outpulsed on the trunk after the End-to-End Signaling Delay (EESD) timer expires. If the far end is not ready for an incoming call, the call will fail because no dial tone will be detected by the system.

The recall transfer for applications, such as Meridian Mail, Customer Controlled Routing or Meridian Link, is not supported.

Optimization is not performed if a Conference key is used.

AC15 trunks using MFC signaling are not supported.

When a trunk has been split, the Release Key functions as a Hold Key. A user cannot selectively release one call in a split mode.

With new functionality of the Release Key, the following events occur if party Z goes on-hook when a trunk is split:

- if HCC = NO, the active call is put on hold;
- if HCC = YES, all calls are released and party X and party Y are connected; or
- if HCC = XFER, or if one of the calls is active, the trunk is released and party X and party Y are connected. If both calls are held, then there is no effect.

Feature interactions

AC15 Recall: Transfer from Norstar

If a recall message is received on a “split out” AC15 trunk, then this message is ignored.

Authorization Codes

Authorization Codes, Basic Authorization Codes and Station Specific Authorization Codes are not supported with the ACRL feature. Recall digits are outpulsed with the End-to-End Signaling, which does not support the aforementioned features. If a user has trunk access restrictions, it is not possible to override the priority by dialing an authorization code. Another trunk will be seized.

Autodial
Last Number Redial

Autodial and Last Number Redial are supported with the AC15 Recall: Transfer from Meridian 1 on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone detection is performed by the system.

Additional transfers are supported if the stored digits are outpulsed without any treatment. For example, a route is seized and the route access code is outpulsed to the far end and interpreted as a Directory Number. No dial tone detector or timer is started, so the digits are outpulsed immediately without checking the state at the far end.

Call Park

If party Z parks the call initiated by party X (an external caller), then the AC15 Recall: Transfer from Meridian 1 cannot be used to call party Y. Party Z can neither park, selectively, one member of a split trunk nor park a whole split trunk. This avoids a recall to an attendant on the recall originating node that would not be able to send a recall to toggle from one party to another.

Call Detail Recording

Call Detail Recording generates one N record. This record contains information on the first call associated with the Directory Number. Information on the transfer is not retained.

Conference

The use of the Conference key does not activate the AC15 Recall: Transfer from Meridian 1 feature. Conference call is not supported because it is not possible to have two parties on the same trunk.

Redirection

If party Z transfers party X to party Y through Call Forward/Hunting, then the AC15 trunk to party Y is not supported. The AC15 trunk cannot be split. If possible, another AC15 trunk is used.

Set Display

The toggling from party X to party Y changes on the display of party Z. All digits dialed during the call are displayed. If Party X or Party Y goes on-hook, party Z still displays the number dialed. If an additional extension is dialed, the digits are added to the previously dialed digits.

Speed Call
Network Speed Call

Speed Call and Network Speed Call are supported with the AC15 Recall: Transfer from Meridian 1 on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone is detected by the system.

Additional transfers are supported if the digits are outpulsed without any treatment. For example, the route access code will be outpulsed to the far end. No dial tone detector is assigned and no timer is started so the digits are outpulsed immediately without checking the state at the far end.

Feature packaging

AC15 Recall: Transfer from Meridian 1 requires the following packages:

- AC15 Recall (ACRL) package 236
- International Supplementary (SUPP) package 131
- UK Program (UK) package 190
- Autodial Tandem Transfer (ATX) package 258

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Disable the End-to-End Signaling Tone to originating party at the EEST prompt.

2 LD 16 – Define the route accepting recall signal.

3 LD 14 – Define the AC15 trunk.

4 LD 11 – Define the Aries sets.

LD 15 – Disable the End-to-End Signaling Tone to originating party at the EEST prompt.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Customer Features and Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
EEST	NO	End-to-End Signaling Tone to originating party.

LD 16 – Define the route accepting recall signal.

Prompt	Response	Description
REQ	NEW CHG	New, or Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
...		
ROUT	0-511 0-127	Route Number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE	Trunk type requires response when REQ = NEW.
...		
CNTL	YES	Changes to controls or timers.
TIMR	EESD 0 - (1024) - 4992	End-to-End Signaling Delay timer. If EESD = 0, the timer is not started and the buffered digits will not be outpulsed.
...		
DLTN	YES	Dial tone provided by the system to the far end switch.
TRRL	YES	Recall signal can be received and transmitted on this route.

LD 14 – Define the AC15 trunk.

Prompt	Response	Description
REQ	NEW CHG	New, or Change.
TYPE	TIE	Type of trunk.
...		
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XFEM	Extended Flexible E & M trunk card.
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
SIGL	WR4	AC15 signaling.
SUPN	YES	Answer and disconnect supervision required.
CLS	DTN	Digitone Class of Service.

LD 11 – Define the Aries sets.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.
TYPE:	xxxx	Telephone type where: xxxx = 2006, 2008, 2016, 2216 and 2616.
...		
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		

KEY	0-69 TRN 0-69 NUL	To add or remove a Call Transfer key.

Feature operation

1. Party X initiates a call to Party Z through an AC15 circuit.
2. Party Z places Party X on hold and calls Party Y through the same AC15 circuit.
3. Party Z completes the transfer. The AC15 circuit between Party X and Party Z is released.

AC15 Recall: Transfer from Norstar

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Feature description

The AC15 Recall: Transfer from Norstar (TRRL) feature is typically used in network configurations where a great number of branch nodes (small offices using a Norstar key system) are linked to a centralized system with back office functions (for example, answering facilities, Public Switched Telephone Network access) using AC15 TIE trunks. With this feature, when the system receives a call that has been transferred from Norstar, it can reuse the same AC15 circuit during the three-party phase and can release it when the transfer is complete. Therefore, call blocking between the Norstar and the system is reduced, and the number of necessary AC15 trunks could potentially be reduced.

A call between Party X (on the system) and Party Z (on Norstar) is established. Party Z invokes the transfer feature on Norstar. A recall signal (similar to a dial pulse 1) is sent over the AC15 trunk, which is detected by the Extended Flexible E&M pack (XFEM) card on the system. Party X is placed on hold by the system, dialtone is provided to Party Z, and dialed digits are expected on the trunk. A new call to Party Y progresses based on the analysis of the received digits. Subsequent recall signals are used by Party Z to toggle between Party X (the original party) and Party Y (the desired party).

During the three-party phase, if the active party (X or Y) disconnects, dialtone is provided to Norstar. If the held party (X or Y) disconnects, the active call is unaffected. In both cases, the AC15 trunk is not disconnected. When Party Z goes on-hook, a release signal is received, the transfer is completed, and the AC15 trunk is released. If the transfer cannot be completed due to access restrictions, the Access Denied (ACCD in LD 15) intercept treatment is provided to the held party and the active party is disconnected. If the transfer cannot be completed because the active call is not in a ringing or established state, the active call is abandoned and the held party recalls the attendant. During the three-party phase, only one trunk is used. Without this feature, however, two AC15 trunks are needed.

Operating parameters

There is no signaling capability to inform Norstar that the second called party (Party Y) has answered. Similarly, there is no signaling capability to inform Norstar that there has been a release of any call by Party X or Party Y on the line.

This feature enables the system to process a recall signal received on the AC15 trunk. It does not enable the system to send such a signal.

The AC15 trunk must be configured with digitone Class of Service and answer supervision.

Currently, only Norstar key systems are supported on the far end.

When dialtone is provided by the system, the digits are dialed according to the system's numbering plan, not that of the Norstar.

This feature requires the XFEM trunk card (NT5K19AC) or later. It is only applicable to the UK market.

Whenever a recall signal from Norstar is not allowed by the system (for example, impossible to put a call on hold, conference, or transfer chaining prevention), the signal is ignored.

Feature interactions

This feature introduces a new concept: a trunk can now put a call on hold and perform a transfer. Wherever possible, treatment is kept consistent with that of an analog (500/2500 type) telephone performing the same actions.

AC15 Recall: Timed Reminder Recall

A transfer performed by an AC15 trunk using the Transfer from Norstar feature to another AC15 trunk is not subject to Timed Reminder Recall. This is to prevent a call transferred by someone on Norstar from recalling the system's attendant.

It is recommended that all AC15 cards on the network's system are NT5K19AC or later. This is mandatory for the system which directly interface with the Norstar. This requirement applies to all AC15 cards for this switch, including the cards that do not directly interface with the Norstar.

AC15 Recall: Transfer from Meridian 1

If a recall message is received on a "split out" AC15 trunk, then this message is ignored.

Attendant Consoles

If a party dials the DN of an attendant, current operation interprets this as an attendant recall request. The call is presented to the attendant on the ICI RLL. If the attendant answers, the transferred party is on the source and the controlling party is on the destination. If enhanced secrecy is denied, a three-party conference is established between the transferred party, the controlling party and the attendant.

With the Transfer from Norstar feature, if Y is an attendant it is a simple call presented on the source side of the attendant. When the attendant answers, a two-party conversation is established between the party and the attendant. No conference is established. To prevent transfer chaining, the attendant cannot transfer this party to another destination – dialed digits will be ignored.

Break-in to Enquiry Calls

It is not possible to Break-in to an enquiry call made by the Transfer from Norstar feature.

Call Detail Recording

In all cases, the conditions required for generating a CDR record are not changed by this feature. If the customer wants to see all records generated with this feature, the route containing the AC15 trunk must be configured with CDR = YES. If the customer only wants to see records generated as if the call were transferred by a local set, the route containing the AC15 trunk must be configured with CDR = NO.

It is possible to generate S records during simple call transfers. In multiple call transfers, X records are produced in some situations due to the CDR Enhancement feature.

It is possible, with this feature, to define an initial connection record (Q record) for incoming calls. The Q record is generated when an incoming trunk and an ACD agent are connected.

The CDR with Outpulsed Digits and the CDR Time to Answer features can also be applied to this feature.

Call Park

Remote access to Call Park from AC15 TIE trunks is not permitted. It is not possible to park an AC15 trunk if it has a call on hold. When an AC15 trunk is parked, it is not allowed to initiate a consultation call.

Call Trace

When the AC15 trunk is handling two calls during the three-party phase, both calls are traced in LD 80.

Call Trace Enhancement

This enhancement is applicable to theAC15 Recall: Transfer from Norstar feature. A record is issued any time the call state or the active call changes after a recall or a release message has been received from Norstar.

Calling Party Control

If a call comes from a trunk with calling party control, and the destination is a trunk, transferring the call is not allowed. When the AC15 trunk receives the release message, Access Denied treatment is provided.

Call Transfer

A party involved in a consultation call (an active or held party) cannot initiate a consultation call for preventing call chaining. This principle is maintained in the following cases:

- the party is an AC15 trunk (if it attempts to initiate a consultation call, the recall signal is ignored), and
- the party is a local set, but the consultation call is made by an AC15 trunk.

Conference

It is not possible in any situation with Transfer from Norstar to establish a three-party conference. It is not possible for an AC15 trunk to initiate a consultation if it is involved in a conference.

Dial Access to Group Calls

If Norstar sends a recall signal in order to initiate a consultation, the consultation will not be authorized because it is not possible to put a group call on hold. It is, however, possible to transfer a party to a group call using an AC15 trunk.

Digital Private Network Signaling System 1 (DPNSS1) Route Optimization

If the call is the active call at the originating exchange and the originator (including an AC15 trunk) has another call on hold, Route Optimization will not be initiated.

If the call is the active call at the terminating exchange and the terminator (including an AC15 trunk) has another call on hold, Route Optimization will not be initiated.

If the call is held at the originating exchange (including an AC15 trunk), Route Optimization should not be initiated. When this call is restored as the active call, it may be optimized.

If the call is held at the terminating exchange (including an AC15 trunk), Route Optimization may be requested by the originator, but the terminating circuit switched network will reject it. When this call is restored as the active call, it may be optimized.

If the call has been transferred to an already answered party (including an AC15 trunk), the transfer signaling sequence is used to initiate optimization.

During a route optimization attempt, if an AC15 trunk is involved in the call either at the originating or terminating exchange, a recall signal is ignored.

DPNSS1 Three-party Service

When the set on Norstar completes a call transfer between two sets located within a DPNSS1 network:

- DPNSS1 access restriction are checked
- the set's displays are updated, and
- DPNSS1 route optimization after transfer can be activated.

Incoming Call Indicator Enhancement

If the held party recalls the attendant due to intercept or recall treatment, the recall is presented to the corresponding ICI key (INT or RLL).

Initialize

If initialization occurs during the three-party phase, the call on hold is cleared. If the active call is established, it is kept, otherwise it is cleared as well (and the AC15 trunk is idled).

MFC Signaling

AC15 trunks using MFC signaling are not supported.

Music

A party put on hold by an AC15 trunk hears music if Music is configured.

Periodic Pulse Metering

If Party Z (on Norstar) calls Party X and transfers the call to Party Y, if Party X is an outgoing trunk with PPM or Advice of Charge on the system, the call is charged against the AC15 trunk route's meter until the transfer is completed. When Party Z completes the transfer in ringing status, the charges still accumulate in the AC15 trunk route's meter. If the call is in established status, the charges accumulate against Party Y, if Party Y has a meter, or otherwise against the customer meter.

Radio Paging

It is possible for an AC15 trunk to complete a transfer to a paging trunk. If the held party is a trunk and the RPA recall timer is configured, the call recalls the attendant when the timer expires.

A set (or Attendant Console) involved in a consultation call cannot pick up (by the RPAN Flexible Feature Code) a paged call which is itself a consultation call. This principle applies to consultation calls made with AC15 trunks.

Slow Answer Recall for Transferred External Trunks

In both standalone and Network Attendant Service (NAS) environments, when a call is transferred to a ringing set on the system by an AC15 trunk, the RTIM recall timer is not started.

Feature packaging

The AC15 Recall (ACRL) package 236 must be equipped to activate the Transfer from Norstar feature.

For recalls to the Night DN, International Supplementary Features (SUPP) package 131 is required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Define the access denied intercept treatment.

2 LD 16 – Define the route accepting recall signal.

3 LD 14 – Define an AC15 TIE trunk.

LD 15 – Define the access denied intercept treatment.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	INT	Intercept treatment options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
INTR	YES	Intercept treatment.
- ACCD	(OVF ATN ATN ATN)	Choice of access denied intercept treatment.
- LLT	(OVF) OFA ATN	Treatment given to calling party when dialtone timer expires: when OVF or OFA is entered, overflow is provided. When ATN is entered, the party is forwarded to the attendant.

LD 16 – Define the route accepting recall signal.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE	Trunk type.
NEDC	ETH	Near-end disconnect control by either originator or terminator.
FEDC	ETH	Far-end disconnect control by either originator or terminator.
DLTN	YES	Dialtone provided by the system to the far-end switch.
TRRL	YES	AC15 Recall: Transfer from Norstar. An AC15 trunk on this route is able to receive a recall signal. Prompted with ACRL package 236 if TKTP = TIE, and DTRK = NO.

LD 14 – Define an AC15 TIE trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE	TIE trunk.
TN	 l s c u c u	Terminal number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XFEM	XFEM card.
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
SIGL	WR4	AC15 signaling.
SUPN	YES	Answer and disconnect supervision.
CLS	DTN	Digitone Class of Service.

Feature operation

Initiate a consultation

A call is established between Party X (a set or trunk) on the system and Party Z on Norstar, through an AC15 trunk. When Party Z initiates a consultation call (Norstar sends a recall signal), Party X is placed on hold and dialtone is provided to Norstar using the same AC15 trunk. The digits received from Norstar are processed according to the system's dialing plan, and eventually Party Y (a set, trunk, or Attendant Console) rings. If no digits are received from Norstar for 14 seconds while Z hears dialtone, overflow tone (for 14 seconds), and then silence (indefinitely) are provided to Party Z. At any time, Norstar may then send another recall signal to be reconnected to Party X.

Toggling during the three-party phase

If Z toggles (Norstar sends a recall signal) while calls with both X and Y are established, the active party is put on hold, and the held party becomes active. If the active call is not established (for example, dialing, ringing, or busy), it is disconnected.

If the held party has released, then if the active call is established it is put on hold, otherwise it is disconnected; in both cases dialtone is provided to Z.

Active or held party disconnects during the three-party phase

If the active party (X or Y) disconnects during the three-party phase, dialtone is provided to Z and a new call can be processed. If the party on hold (X or Y) disconnects, the active call is unaffected. In both cases, the AC15 trunk is not disconnected.

Complete the transfer

Party Z completes the transfer from X to Y, regardless of which is the active party, by going on-hook (Norstar sends a release signal). The transfer is allowed when the active call is ringing or established. Note that if Y is a trunk, although Z is hearing ringback tone, the call will not be considered in a proper state for being transferred until Y's end of dialing timer (EOD or ODT) has expired, or Z has pressed the # sign. In the other call states, the active call is abandoned and the held party recalls the attendant. If the call cannot be transferred due to access restrictions, the active party is disconnected and the held party is given the Access Denied (ACCD) intercept treatment. In all cases, when the release signal is received, the AC15 trunk is disconnected.

Access Restrictions

Contents

This section contains information on the following topics:

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Feature description

Access Restrictions limit terminal access to the exchange network, private network, and certain services and features.

Access Restrictions can be temporarily overridden by the use of other features, if equipped, including Forced Charge Account, Authorization Code, and System Speed Call.

During the call origination process, access checks are made by the system on the following:

- the Class of Service (CLS) of the individual terminal
- the Trunk Group Access Restriction (TGAR) code of the terminal if a direct trunk access code is dialed or as an optional feature when a Basic Alternate Route Selection (BARS) or Network Alternate Route Selection (NARS) access code is dialed
- the area and exchange codes dialed by terminals with Toll Denied or Conditionally Toll Denied Class of Service using direct trunk access codes and Code Restriction tables, and
- the Network Class of Service (NCOS) of the terminal if BARS/NARS or Coordinated Dialing Plan (CDP) access codes are dialed or if direct trunk access codes are dialed and New Flexible Code Restriction tables are programmed.

If any restrictions are detected when a call is placed, the call is given intercept treatment as defined in the Customer Data Block.

Class of Service restrictions

The Class of Service restrictions assigned to telephones and TIE trunks control the degree of access to and from external networks and certain features within the system. The eight possible Class of Service Access Restrictions are described in this feature module. These restrictions are applied by service change overlay programs to terminals. Table lists the type of terminals and the corresponding overlay program.

Table 2
Type of terminal and the corresponding overlay program for configuring Class of Service restrictions.

Terminal Type	Overlay
Analog (500/2500 type) telephone	10
Meridian 1 proprietary telephones, Meridian Mail channels	11
Incoming TIE trunks	14
Authorization Codes	88
DISA ports	24

Descriptions of the eight Class of Service Access Restrictions follow, from the most restricted to the least restricted.

Fully Restricted Service

There are three levels of Fully Restricted Service:

- FR2
  - — allowed to originate and receive internal calls
  - — denied access to TIE and Common Controlled Switching Arrangement networks
  - — denied access to and from the exchange network, either by dialing, through an attendant, or using call modification from an unrestricted telephone

    Call modification takes place when certain features are activated while a call is in progress (for example, Call Park, Call Pickup, Call Transfer, Conference, or Night Answer).
- FR1
  - — allowed to originate and receive internal calls

— allowed access to TIE and CCSA networks

— denied access to and from the exchange network, either by dialing through an attendant or by using call modification from an unrestricted telephone

Note: In a networking environment, incoming and outgoing calls can be extended, through call modification, to a telephone with CLS = FR1.

If a telephone with CLS = FR1 is in a Multiple Appearance DN (MADN) arrangement, the call may be presented if at least one of the telephones has CLS = UNR. Once the call is presented, it will ring all telephones in the MADN group. However, only UNR telephones can answer the call.

- FRE
  - — allowed to originate and receive internal calls
  - — allowed access to TIE and CCSA networks
  - — allowed access to and from the exchange network using call modification from an unrestricted telephone
- denied access (either by dialing or through an attendant) to and from the exchange network

Note: The FRPT prompt in LD 17 allows or denies access to incoming calls for FRE CLS telephones. It allows FRE calls to Call Pickup, Night Answer, and to receive modified calls.

The assignment of Incoming Call Indicator (ICI) keys allows the attendant to recognize which calls are fully restricted:

- DF0 = calls from FRE, FR1, and FR2 CLS, and
- DL0 = calls from CUN, CTD, TLD, SRE, and UNR CLS.

Semi-Restricted Service (SRE)

- allowed to receive calls from the exchange network
- restricted from all dial access to the exchange network
- allowed to access the exchange network through an attendant or an unrestricted telephone only

Toll Denied Service (TLD)

- allowed to receive calls from the exchange network
- allowed access to WATS trunks for toll calls using direct trunk access codes, unless New Flexible Code Restriction (NFCR) is programmed to deny certain digits
- denied from calls on Central Office/Foreign Exchange (CO/FX) trunks where 0 or 1 is dialed as a first or second digit following a direct trunk access code. Special numbers, such as 411, 611, and 911, are allowed by default unless restricted specifically by NFCR.
- denied from toll calls on CO/FX trunks when BARS or NARS access codes are dialed, unless NFCR tables allow the call
- allowed toll calls on WATS trunks using BARS or NARS access codes, unless NFCR tables deny digits
- allowed access to the toll exchange network through an attendant or an unrestricted telephone
- allowed toll calls and special number calls on TIE trunks, unless NFCR tables specifically deny certain digits. Direct trunk access to toll calls on TIE trunks is permitted, as well as BARS or NARS access.

Conditionally Toll Denied Service (CTD)

- allowed to receive calls from the exchange network
- allowed access to WATS trunks for toll calls using direct trunk access codes, unless New Flexible Code Restriction (NFCR) is programmed to deny certain digits
- denied from calls on CO/FX trunks where 0 or 1 is dialed as a first or second digit following a direct trunk access code (special numbers excepted). New Flexible Code Restriction tables can be used to deny or allow certain calls on these routes.

- allowed access to toll calls on CO/FX/WATS trunks placed using BARS or NARS or CDP access codes. NFCR tables, if programmed on the routes, are ignored for CTD users dialing Electronic Switched Network (ESN) access codes.
- allowed toll calls and special number calls on TIE trunks, unless NFCR tables specifically deny certain digits. Direct trunk access is permitted as well as BARS or NARS access. NFCR tables deny calls for these users only if direct TIE trunk access codes are used.

Conditionally Unrestricted Service (CUN)

- allowed access for calls placed through Automatic Number Identification (ANI) trunks
- denied access for all other types of outgoing calls

Unrestricted Service (UNR)

- allowed to originate and receive calls from the exchange network

The eight possible Class of Service Access Restrictions are described in Table 3.

Table 3
Class of Service Access Restrictions chart

	UNR	CTD/CUN	TLD	SRE	FRE	FR1	FR2
Incoming trunk calls	Yes	Yes	Yes	Yes	No Yes, if using call modification. (See page 150)	No	No
Outgoing non-toll trunk calls	Yes	Yes	Yes	No direct access Yes, if using attendant or UNR telephone	No direct access Yes, if using UNR telephone	No	No
Outgoing toll trunk calls (0 or 1+ on COT or FX)	Yes	No direct access Yes, if using BARS/ NARS	No direct access Yes, if using attendant or UNR telephone	No direct access Yes, if using attendant or UNR telephone	No direct access Yes, if using UNR telephone	No	No
To/From TIE trunk	Yes	Yes	Yes	Yes	Yes	Yes	No
To/From internal	Yes	Yes	Yes	Yes	Yes	Yes	Yes
BARS/ NARS calls TGAR = No	Uses NCOS only	Uses NCOS only	Uses NCOS and CLS	Uses NCOS and CLS	Uses NCOS and CLS	Uses NCOS and CLS	Uses NCOS and CLS
BARS/ NARS calls TGAR = Yes	Uses NCOS and TGAR	Uses NCOS and TGAR	Uses NCOS, CLS, and TGAR	Uses NCOS, CLS, and TGAR	Uses NCOS, CLS, and TGAR	Uses NCOS, CLS, and TGAR	Uses CLS only

Code Restriction

Code Restriction allows limited access to the toll exchange network to stations and TIE trunks with a Toll Denied Class of Service (TLD). A Code Restriction Block that specifies the allowed area and exchange codes (200 through 999) is built for each trunk route. This block restricts access to specific area and exchange codes by monitoring the digits dialed.

There can be only one Code Restriction Block per route. The only routes that use Code Restriction Blocks are Central Office Trunk (COT) and FX, since they are toll routes. Code Restriction Blocks are ignored for all other types of routes.

When a telephone or TIE trunk with a CTD, CUN, or TLD Class of Service directly access a COT or FX route, the system examines the Code Restriction Block to determine the call eligibility.

Special numbers 01, 011, 411, 611, 800, and 911 are allowed by default. These special numbers, however, can be restricted in the Code Restriction Block so that they cannot be dialed successfully.

Code Restriction Blocks only perform three-digit screening. For 1+ dialing areas, the system can ignore the 1 when examining the TLD telephone dialed number. The 1 is later outpulsed with the dialed number to complete the call successfully.

Trunk Group Access Restriction

Trunk Group Access Restriction (TGAR) controls access to the exchange network, TIE trunks, CCSA trunks, and paging and dictation services.

Telephones (LD 10, LD 11), TIE trunks (LD 14), Direct Inward System Access (DISA) trunks (LD 24), Meridian Mail channels (LD 11), and Authorization Codes (LD 88) are assigned a TGAR code, which is used to block access to certain trunk groups entirely.

There can be up to 32 TGAR codes in use on a system (0-31).

When a telephone or TIE trunk dials the access code to a trunk route, the system first checks the Class of Service of the terminal. If access is allowed, the TGAR is checked next. If the TGAR of the originating terminal matches one of the listed Trunk Access Restriction Group (TARG) codes programmed against the trunk group, access is denied. Intercept treatment is given to denied calls. A list of TARG codes can be programmed in LD 16 against each route, where applicable, to block access by certain terminals.

Optionally, the TGAR can be used to block access to certain routes even when a BARS or NARS access code is dialed and the route is being seized. To enable/disable the TGAR option, the TGAR prompt must be defined in the Electronic Switched Network (ESN) data block in LD 86.

When denied access because of TGAR, a user may still gain access to a route through the Attendant Console or an unrestricted terminal.

If the attendant uses the Trunk Group Busy (TGB) keys on the console to make trunk groups busy, terminals with TGAR code 0-7 are intercepted to the attendant when they access the route by dialing or try to gain access using ESN access codes. Terminals with TGAR code 8-31 continue to have access to the route, unaffected by the activation of the TGB keys.

The default, TGAR code 1, means the terminal is Conditionally Toll Denied (CTD).

The following example further explains Trunk Group Access Restrictions. Assume a customer has seven trunk routes:

TGAR	**Access denied to routes**
Route 0	COT
1	WATS
2	FX 1
3	FX 2
4	TIE 1
5	TIE 2
6	Paging

Assume the following seven TGAR codes are required:

TGAR	Access denied to routes
0	No restrictions
1	0, 1, 2, 3, 4, 5, 6 (default)
2	2, 3, 4, 5
3	3, 4, 5
4	2, 6
5	3, 4, 5, 6
6	5, 6

The TGAR/TARG matrix summary is as follows:

Trunk Type	Route number	TARG Code
		0 1 2 3 4 5 6 7-31
COT	0	1
WATS	1	1
FX 1	2	1 2 4
FX 2	3	1 2 3 5
TIE 1	4	1 2 3 5
TIE 2	5	1 2 3 5 6
Paging	6	1 4 5 6

It follows from the matrix summary that a telephone or TIE trunk was assigned one of the following TGAR codes:

- 0 (has no restrictions)
- 1 (cannot access trunk routes 0 through 6)
- 2 (cannot access trunk routes 2 through 5)

- 3 (cannot access trunk routes 3 through 5)
- 4 (cannot access trunk routes 2 and 6)
- 5 (cannot access trunk routes 3 through 6)
- 6 (cannot access trunk routes 5 and 6)

Trunk signaling arrangements

Trunk-to-trunk connections are further controlled by the signaling and supervision arrangements assigned to each trunk. Table 4 summarizes the trunk signaling arrangements.

Table 4
Trunk signaling arrangements

From	To		
	Trunk with/ without disconnect supervision	Paging dictation trunk	Telephone (non-trunk)
Trunk with disconnect supervision	Yes	No	Yes
Trunk without disconnect supervision	No	No	Yes
RAN/Paging dictation trunk	No	No	No
Telephone	Yes	Yes	Yes
Note: Yes = connection allowed No = connection disallowed			

Two outgoing trunks cannot be connected unless a supervising party, local to the system, is conferenced in the call. This is true regardless of the supervisions.

Transfer from a supervised trunk to a non-supervised loop start trunk is not permitted.

Operating parameters

If a conflict exists between the Class of Service (CLS) and Trunk Group Access Restrictions (TGAR), the access denied restriction takes precedence.

Access Restrictions are applied through service change overlay programs. Access to telephone and trunk features is denied in the respective data block by allowing the system to default to a denial, by not entering the appropriate feature code, or by not assigning the feature to a key/lamp pair. You must enable the features and Access Restrictions you want, on a customer and telephone level.

Services such as paging and dictation can be restricted through TGAR codes, because the auxiliary equipment is linked to the system by way of trunks.

Feature interactions

AC15 Recall: Timed Reminder Recall

With call modification, a trunk-to-trunk connection is controlled by signaling, recall capability and the supervision assigned to each trunk. For example, an established call from an unsupervised trunk cannot be transferred over another trunk.

When the AC15 Timed Reminder Recall feature is to be activated, an established call with an unsupervised trunk may be extended over an AC15 trunk because the connection is controlled before the called party answers by the AC15 recall timer.

Call Park

A call can be parked on any DN, regardless of its Class of Service. Access to a parked call is governed by the same Class of Service restrictions for normal trunk-to-telephone call processing. Table 5 details the restrictions. These restrictions can be overridden with the Authorization Code.

Table 5
Parked call Access Restrictions.

Parked call type	Accessing telephone Class of Service		
	FRE	FR1	FR2
Telephone	allowed	allowed	allowed
CO/FX/WATS	denied	denied	denied
DID Trunk	denied	denied	denied
TIE trunk	allowed	allowed	denied

Call Pickup Network Wide

All Access Restrictions applicable to Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS) calls (including Class of Service, Network Class of Service, Trunk Barring (TBAR), and New Flexible Code Restriction (NFCR) restrictions based on digit manipulation) apply to a redirected call from the receiving node to the requesting node. This means that there are no limitations added to the access restriction checks for calls being redirected by the Call Pickup Network Wide feature.

If the call is blocked because of any of these Access Restrictions on either the receiving, tandeming, or requesting node, the originally called party is re-rung and the party attempting to pick up the call receives overflow tone.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The connection between the network user (extension or trunk) and the DPNSS1 UDP trunk can be barred based on the Class of Service Restrictions of the parties involved. The connection between the network user (extension or trunk) and the DPNSS1 trunk can also be barred based on the Trunk Group Access Restrictions feature. It is possible to bar the connection between originator and terminator through a DPNSS1 UDP trunk based on the DPNSS1 signaling information.

The Code Restriction sub-feature is not supported.

Direct Inward System Access

Access Restrictions are assigned to the Direct Inward System Access (DISA) DN as they are to any station within the system. Separate Access Restrictions are also assigned to authorization codes used by DISA callers.

Group Hunt

If a routing-associated DN is programmed in a group hunt list, the Access Restrictions based on the Class of Service and/or TGAR of the calling station/ route apply.

ISDN QSIG/EuroISDN Call Completion

ISDN QSIG/EuroISDN Call Completion does not override Access, Call Restriction or Trunk Group Access Restrictions. When Call Completion is activated, the second call has the same restrictions as the initial call that received either no answer or a busy indication.

New Flexible Code Restriction

The Code Restriction feature and New Flexible Code Restriction cannot be implemented simultaneously for the same customer.

Scheduled Access Restrictions

The Trunk Access Restriction Group (TARG) defined for each route is not altered by Scheduled Access Restrictions. Access to the route is denied to any telephone or trunk assigned a Trunk Group Access Restriction code that is part of the TARG.

Trunk Barring

Trunk Barring is at the top of the hierarchy for Access Restrictions.

Virtual Network Services

Any VNS call is subject to the same Class of Service restrictions as if the call was performed on a TIE trunk, regardless of the type of Bearer trunk used.

Trunk Group Access Restrictions (TGARs) do not apply to VNS, and therefore they never restrict a VNS call from being made.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Define a Class of Service and TGAR code for analog (500/2500 type) telephones.

2 LD 11 – Define a Class of Service and TGAR code for proprietary telephones.

3 LD 14 – Define a Class of Service and TGAR code for trunks.

4 LD 88 – Assign a Class of Service to the Authorization Code classcode.

5 LD 86 – Enable or disable the Trunk Group Access Restriction (TGAR) option.

6 LD 24 – Assign a Class of Service to Direct Inward System Access (DISA) numbers.

7 LD 17 – Allow or deny incoming calls to telephones with the FRE Class of Service for all customers.

8 LD 16 – Add or change the TARG code for a trunk route.

9 LD 19 – Implement Code Restriction on trunk routes.

10 LD 16 – Define toll access digits that are to be ignored for Code Restriction.

LD 10 – Define a Class of Service and TGAR code for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.

LD 11 – Define a Class of Service and TGAR code for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.

LD 14 – Define a Class of Service and TGAR code for trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	TIE	TIE trunk.
	ISA	Integrated Services Access trunk.
	CSA	Common Control Management Access Line.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
TGAR	0-(1)-31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.
	X	Precede with X to remove
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.

LD 88 – Assign a Class of Service to the Authorization Code classcode.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	AUB	Authcode Data Block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (see LD 15 for description).
CLAS	0-115	Classcode number.
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
NCOS	(0)-99	Toll Restricted.

LD 86 – Enable or disable the Trunk Group Access Restriction (TGAR) option.

Prompt	Response	Description
REQ	CHG	Change.
CUST	xx	Customer number, as defined in LD 15
FEAT	ESN	Electronic Switched Network.
...		
TGAR	(NO) YES	Do not check for Trunk Group Access Restrictions when a call is placed through BARS. Check for Trunk Group Access Restrictions when a call is placed through BARS.

LD 24 – Assign a Class of Service to Direct Inward System Access (DISA) numbers.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DIS	Direct Inward System data block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (see LD 15 for description).
DN	xxx...x	DISA Directory Number.
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
NCOS	(0)-99	Network Class of Service.
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.

LD 17 – Allow or deny incoming calls to telephones with the FRE Class of Service for all customers.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters.
...		
FRPT	(NEFR) OLFR	(Deny) allow incoming trunk calls to telephones with FRE CLS, using call modification.

LD 16 – Add or change the TARG code for a trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TARG	1 2 3…31	Route TARG codes (list each TGAR to be blocked from using this route – put a space between each entry). To remove an entry, precede with X.

LD 19 – Implement Code Restriction on trunk routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CRB	Code Restriction Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127 	Route number of COT or FX For Large Systems For Small Systems and Succession 1000 systems There can be only one Code Restriction Block for each COT or FX route.
CLR	ALOW	Allow all NPA/NXX codes except those entered in response to the prompt DENY.
	DENY	Deny all NPA/NXX codes except those entered in response to the prompt ALOW.
	<CR>	Used when REQ = CHG.
ALOW	xxx xxx…	If CLR = DENY, enter the NPA/NXX codes (200-999) allowed.

DENY	xxx xxx…	If CLR = ALOW, enter the NPA/NXX codes (200-999) denied.

LD 16 – Define toll access digits that are to be ignored for Code Restriction.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
OABS	x x x	Outgoing digits (0-9) to be ignored.

Feature operation

No specific operating procedures are required to use this feature.

Activity Codes for Not Ready State

Contents

This section contains information on the following topics:

Feature description 169
Operating parameters 169
Feature interactions 170
Feature packaging 171
Feature implementation 171
Feature operation 173

Feature description

The Activity Codes for Not Ready State feature allows an agent to use the existing Activity Code key to record activities while in the Not Ready State.

Operating parameters

This feature is designed for proprietary sets with display. This feature is not supported for analog (500/2500 type) sets.

The Not Ready State is automatically invoked if the supervisor uses the following keys:

- Observe Agent
- Call Agent
- Answer Agent
- Answer Emergency

Note: When these keys are used, the Activity Code key lamp does not flash.

The Activity entry key and Activity key lamp are not affected if the Program key, the Display key, volume up/down, and handsfree keys are used.

If any key other than the Activity, Handsfree Mute, Dial Pad, Display key or Volume Control key is pressed while entering an Activity code, the Activity key lamp turns dark and any code entered is lost.

Activity Codes for the Not Ready State cannot be activated during Walkaway, Logged Out or Make Set Busy states.

An incoming call to the agents Individual Directory Number (IDN) does not interfere with the Activity Code entered, if the entry is completed before answering the call. If the Activity Code entry is not completed before answering an incoming call, the Activity Code is lost.

Feature interactions

Multiple Queue Assignment

If Multiple Queue Assignment (MQA) is in use, the default Activity code sent to the Meridian MAX becomes the default code for the queue of the agent's last call answered. The ACD D defaults back to the last ACD DN the set was logged into.

Return to Queue on No Answer

If a call is not answered by an agent, the call is sent back to the Automatic Call Distribution (ACD) queue and the agent's set is automatically put into the Not Ready State. The Activity key lamp does not flash.

Feature packaging

There are two minimum package combinations required to operate this feature: one for Meridian MAX and the other for the Symposium Call Center.

The feature packaging requirements for Meridian MAX are:

- Automatic Call Distribution, Account Code (ACNT) package 155
- Automatic Call Distribution Package D (ACD D) package 50
- ACD D, Auxiliary Link Processor (LNK) package 51
- Automatic Call Distribution Package D, Auxiliary Security (AUXS) package 114

The feature packaging requirements for Symposium are:

- Automatic Call Distribution, Account Code (ACNT) package 155
- Symposium Call Center (NGCC) package 311

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11– Define an Activity Code key for proprietary sets.

2 LD 23 – Enable Activity Codes in the Not Ready State for an ACD queue.

LD 11– Define an Activity Code key for proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Set type Where a...a = proprietary set with display capabilities (2006, 2008, 2009, 2016, 2018, 2112, 2216, 2616, i2002, i2004, and SL1).
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15.
...	...	
KEY	xx ACNT	xx = Key number (the ACNT key cannot be configured as 0).

LD 23 – Enable Activity Codes in the Not Ready State for an ACD queue.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	ACD	Automatic Call Distribution data block. Requires Basic Automatic Call Distribution (BACD) package 40.
...	...	
ACNT	x...x	Account (Default activity code). Maximum four digits. Prompted if the ADS data block is built and the DCUS (Maximum number of ACD customers) setting in LD 17 is greater than one.
NRAC	YES	Enable Not Ready Activity Codes. (NO) = default.
- NDFL	xxxx	Not Ready Default code. Must be equipped with ACD D or NGCC package.

Feature operation

To enter an Activity Code in the Not Ready State:

1. Press the Not Ready key. The Not Ready key lamp lights and the Activity Code key lamp flashes.
2. Press the Activity key. The Activity key lamp lights steadily.
3. Enter the activity code.

   Note: The * is used to delete one digit at a time. The # symbol delete all the digits entered.

4. Press the Activity key. The activity code is sent to the system and the Activity Code key lamp goes out. This completes the activity code entry.

5 An ACD agent can enter multiple activity codes for each activity completed during any Not Ready Session. Repeat steps 2-4 until all tasks are entered.

6 Press the Not Ready key. The Not Ready key lamp goes out and the agent is placed back into the ACD queue.

To use the Display key in the Not Ready State:

1 The agent presses the Display key. The set display is cleared.

2 The agent presses the Activity key. The previously entered Activity Code appears in the set display.

3 The agent presses the Display key twice (or presses the RLS key) to display the time and date.

Note 1: If an activity code is not entered, the code configured in Overlay 23 (the Not Ready Default code setting) is sent to the system and the Activity Code Key lamp goes out.

Note 2: The ACCT message timestamp is set the first time the Activity key is pressed.

Alarm Management

The Alarm Management feature enhances and updates system operations, administration, and maintenance. Alarm Management provides overall alarm and fault handling, as well as refinements to system displays and alarm processes.

Alarm Management provides the following subfeatures:

- Event Collector
- Event Server
- Alarm Notification
- Escalation and Suppression Thresholds

For information on the Alarm Management feature, refer to "LD 117: Ethernet and Alarm Management" in *Software Input/Output: Administration* (553-3001-311).

Alternative Conference Pad Levels

Contents

This section contains information on the following topics:

Feature description 177
Operating parameters 177
Feature interactions 177
Feature packaging 178
Feature implementation 178
Feature operation 178

Feature description

This feature allows different conference pad levels to be selected during configuration to control the audible levels for parties in a conference call. There are eight acceptable values, from zero to seven.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – The value of the conference pad selection must be specified.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	NET	ISDN and ESN Networking options.
...		
- APAD	x y	Alternative Pad, Where: x = trunk pad selection y = conference pad selection Valid inputs for x are: (0) = default for North America 1 = Australia 2 = UK BPC1031 4-wire TIE trunk 3 = UK BPC902 4-wire TIE trunk 4 = China 5-7 = future usage. Valid inputs for y are: (0) = default for North America 1 = Alternative Conference pads selected The default = 0 when REQ = NEW. The default is the existing value when REQ = CHG. Alternative Conference pads are only provided on specific Conference packs.

Feature operation

No specific operating procedures are required to use this feature.

Alternative Loss Plan for China

Contents

This section contains information on the following topics:

Feature description 179
Operating parameters 180
Feature interactions 180
Feature packaging 180
Feature implementation 180
Feature operation 180

Feature description

This enhancement introduces Alternative Trunk Pad Matrix 4 to be used for China.

At the present time, eight Alternative Trunk Pad Matrix Options are available to satisfy the loss plan requirements of various countries (although only the following five are being used):

0 – Standard, for North America

1 – Australia

2 – United Kingdom

3 – United Kingdom

4 – China

5-7 – Not used

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Modify Customer Data Block to introduce Alternative Pad Matrix 4 for China.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	NET	ISDN and ESN Networking option.
...		
- APAD	 (0) 1 4	Alternative Pad Matrix. 0 = None. 1 = Australia. 4 = China.

Feature operation

No specific operating procedures are required to use this feature.

Alternative Loss Plan

Contents

This section contains information on the following topics:

Feature description 181
Operating parameters 182
Feature interactions 182
Feature packaging 182
Feature implementation 182
Feature operation 184

Feature description

Customers can insert or remove, during administration, an alternative trunk-pad switching matrix using this feature. The loss-plan requirements of different countries can thus be satisfied. The alternative fixed trunk-pad matrix can be used in place of the standard pad switching matrix. Refer to Figure 4 on page 237 for the pad switching matrix.

The customer selects the Alternative Loss Plan (APAD) option in LD 15 to access the alternative matrix. The default option is the use of the standard switching matrix.

The customer selects the Multifrequency Compelled (MFC) Class of Service in LD 14 to switch in the pad in the case of MFC Signaling. The Multifrequency Digit Level is also specified here.

Operating parameters

This feature is not to be used with 1.5 Mbit digital trunks.

Feature interactions

B34 Codec Static Loss Plan Downloading

The alternative loss plan tables must be enlarged as the default table is enlarged.

B34 Dynamic Loss Switching

The alternative loss plan tables must be enlarged as the default table is enlarged.

R2MFC 1.5 Mbps Digital Trunk Interface

Alternative Loss Plan is not supported on 1.5 Mbps DTI.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Configure the Trunks.

2 LD 15 – Configure the Alternative Pad Matrix.

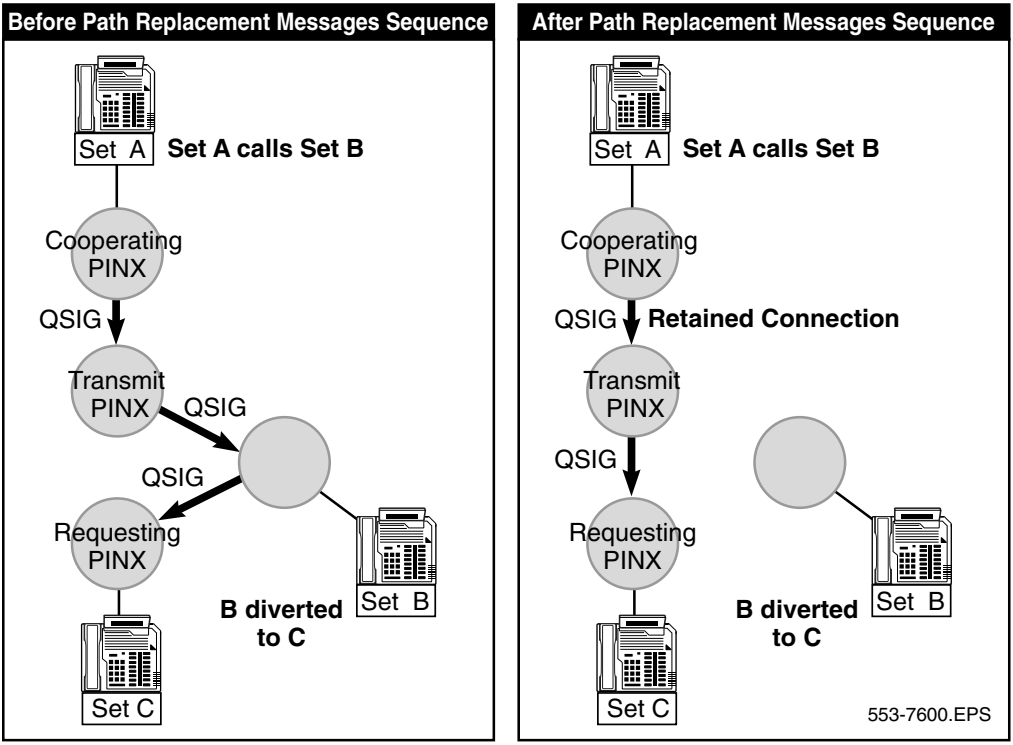
LD 14 – Configure the Trunks.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID TIE	Direct Inward Dial TIE trunk data block.
CLS	MFC	R2 Multifrequency Compelled Signaling.
MFL	(0)-7	Input Multifrequency Digit Level required for signals to the PSTN.
MFPD	(NO) YES	Enter YES for pad in, and NO (the default) for pad out, during MFC signaling.

LD 15 – Configure the Alternative Pad Matrix.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	NET	ISDN and ESN Networking options.
...		
- APAD	(0) 1 (2 - 7)	Alternative Pad Matrix. 0 = None 1 = Australia 4 = China 2, 3 and 5-7 = Future usage (currently set to default)

Figure 4
Alternative Loss Plan pad switching matrix



Feature operation

No specific operating procedures are required to use this feature.

Alternative Routing for DID/DDD

Contents

This section contains information on the following topics:

Feature description 185
Operating parameters 187
Feature interactions 187
Feature packaging 187
Feature implementation 188
Feature operation 188

Feature description

The Alternative Routing for DID/DDD feature provides alternate routing for calls that are recognized as remote Direct Inward Dialing (DID) or Direct Distant Dialing (DDD) Special Numbers (SPN) in a private network. Low cost routing for off-network numbers is also supported.

The Alternative Routing for DID/DDD feature is an enhancement to the Off-net Number Recognition feature.

Refer to *Electronic Switched Network description* (309-3001-100) for further information on the Off-net Number Recognition feature.

For the Alternative Routing for DID/DDD feature, a new type of number is introduced in the SDRR block. It is called the Alternate Routing Remote Number (ARRN). Following each SPN, and only SPNs, a customer can configure ARRNs. For each ARRN, it is also possible to configure an Alternate Route List Index (ARLI).

Call processing follows the same steps as the Off-net Recognition feature follows. The expected digits are compared to the numbers defined in the SDRR Table, and one of the following scenarios applies:

Scenario 1

If a match is found, the following call treatments can occur:

- If the number is recognized as an ARRN, Route Selection with the ARLI defined for the ARRN is performed.
- If the number is in the denied block (such as, SDRR = DENY), standard call blocking takes place.
- If the number is recognized as terminating at the local switch (for instance, SDRR = LDID/LDDD), the call is terminated at the station DN for a DID call, or at the Attendant DN for a DDD call.

Scenario 2

If a match is not found, as in the case of a shorter DN, and the OVLP package is not equipped, timeout handling occurs resulting in call blocking. If timeout handling is not set, call blocking does not occur.

Scenario 3

If a match is not found, as in the case of a shorter DN, and the OVLP package is equipped. The feature then determines if Overlap Sending can be attempted for this call.

- If Overlap Sending is attempted, the timeout handling flag is set to .FALSE (OVLL set to 1).
- If Overlap Sending is not attempted, the timeout handling flag is set to .TRUE (OVLL set to 0)

Note 1: If Overlap Sending is not active the flag can be reset to .FALSE if FNP is equipped and FLEN is a non-zero.

If the number is recognized as terminating at a remote system or Central Office switch (for instance, SDRR = DID/DDD), Route Selection with the RLI that is defined for that SPN is performed.

Route Selection is performed based on the RLI that is found in the table. One RLI corresponds to each SPN. Call processing resumes and the call routes to the Central Office of the terminating Off-net number.

If the route found uses a TIE trunk, then special digit manipulation is applied so that the proper numbers are outpulsed for the call to terminate at the station or attendant.

If the route found does not use a TIE trunk, then the call termination is processed by the current software with digit manipulation, if necessary.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

When Special Numbers (SPN) are used in private network calls, all private network features are supported.

Feature packaging

The Alternative Routing for DID/DDD feature requires Flexible Numbering Plan (FNP) package 160, which depends on the following:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- New Flexible Code Restriction (NFCR) package 49
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59

- Pretranslation (PXLT) package 92
- Incoming Digit Conversion (IDC) package 113
- Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System 1 (DPNSS) package 123
- Digital Access Signaling System 2 (DASS2) package 124

Feature implementation

LD 90 – Assign an ARRN and ARLI to an SPN.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	SPN	Special code translation data block.
LOC	x...x	Location code (3 digits) or extended LOC (3-7 digits). Enter the location code (xxx) and extended code (xxxx) separated by a space.
- RLI	0-255	Respond to the RLI prompt with the Route List Index number from 0-255 (NARS).
...		
- SDRR	ARRN	Respond to the Supplemental Digit Restriction or Recognition prompt with ARRN (Alternate Routing Remote Number).
- - ARRN	x x	Respond to the ARRN prompt with the Alternate Routing Remote Number (up to five digits).
- - - ARLI	0-255	Alternative Route List Index.

Feature operation

No specific operating procedures are required to use this feature.

Application Module Link

The Application Module Link (AML) provides supervisory and control functions for the link that allows host computers and other external processors access to Integrated Services Digital Network (ISDN) network services on the system. The tasks performed by the Application Module Link include link activation, fault detection, maintenance, and traffic reporting. The AML provides the association of telephones with one or more DNs with the host computer. This allows a computer to access basic telephone features of the system. Telemarketing, electronic mail, and other features can take full advantage of ISDN services using the AML.

Application Module

The Application Module (AM), previously known as the Meridian Link Module, is an application processor providing an interface between a host computer and the system providing operations, administration, and maintenance capabilities. It is housed in the Application Equipment Module (AEM). Up to two Application Modules can be put into one AEM chassis in a redundant configuration.

Attendant Administration

Contents

This section contains information on the following topics:

Feature description 193
Operating parameters 195
Feature interactions 196
Feature packaging 199
Feature implementation 199
Feature operation 200

Feature description

Attendant Administration allows the attendant to modify a specific set of features that can be assigned to telephones. The console must have an alphanumeric display, and it must be assigned to the same customer group as the telephones on which the features are to be changed.

Attendant Administration is implemented by assigning a Program key on the flexible feature strip on the Attendant Console. The Program key and a four-digit password allow the attendant to enter the Program mode in a manner equivalent to logging into the system from a system terminal.

When in the Program mode, the Attendant Console key/lamp strip functions are changed from normal call processing to the Attendant Administration programming functions. A plastic overlay is placed over the console key/lamp strips to indicate their programming functions.

The attendant inputs the information by pressing the appropriate key or by entering numbers or letters on the dial pad. The alphanumeric display shows the entered information and provides feedback from the system. The feedback includes the current status of the telephone, the prompts requesting input, and the messages indicating an input error.

The following features can be changed by Attendant Administration (any feature not included in the list cannot be modified or changed by the Attendant Administration feature):

- Call Forward (analog (500/2500 type) telephones only)
- Call Forward Busy (all telephones)
- Call Forward No Answer (all telephones)
- Call Pickup (all telephones)
- Call Pickup Group (all telephones)
- Call Transfer (analog (500/2500 type) telephones only)
- Call Waiting (analog (500/2500 type) telephones only)
- Dial Intercom Group (analog (500/2500 type) telephones only)
- Directory Number (analog (500/2500 type) telephones only)
- Hunt Directory Number (all telephones)
- Hunting (all telephones)
- Last Hunt Key (SL-1 and Meridian digital telephones only)
- Message Waiting (all telephones)
- Permanent Hold (analog (500/2500 type) telephones only)
- Ring Again (analog (500/2500 type) telephones only)
- SL-1 and Meridian digital telephone key assignments
- Speed Calling (analog (500/2500 type) telephones only), and
- Stored Number Redial (analog (500/2500 type) telephones only).

For details on how these features operate, refer to the *Attendant Administration User Guide*.

Operating parameters

Calls cannot be initiated or received by the console while it is in the Program mode.

The attendant can only change data for the customer to which the console belongs.

The system generates Customer Service Change (CSC) messages that indicate changes made to individual telephones. These messages may be output on a system terminal or stored in the History File.

Attempting to change a telephone that is busy is not allowed. A busy telephone is defined as a telephone with any active or held calls or with any active features such as Autodial. There are exceptions. A telephone that has Call Forward All Calls or Make Set Busy activated can be modified.

During the time a telephone is undergoing feature changes by the attendant, it is made Maintenance Busy and is therefore inoperative.

If a console remains idle in the Program mode for 20 minutes, the Program mode is terminated and the console returns to Position Busy.

If an Attendant Console, maintenance telephone, or system terminal tries to log in to the system while another device is logged in, the system displays a message identifying the logged-in device. If a password is then entered, the login is accepted, forcing out the device previously logged in. A console forced out is returned to Position Busy and provided with an output message in the display to indicate what has occurred.

Unlike making service changes at a system terminal, when a Directory Number (DN) is entered for an analog (500/2500 type) telephone that appears elsewhere (as a mixed, Hunt, or Private Line DN), the associated error code (MIX, HUNT, or PVL) is not displayed. If the DN is not valid, an error code is displayed.

The database is automatically dumped during the midnight routine if a transaction has been successfully completed during the previous day. If this datadump fails, the minor alarm lamp on the console will light.

The Attendant Administration password is preserved over an initialization and set to the value on the tape when the system is reloaded.

If the system initializes or reloads while the console is in the Program mode, Attendant Administration is aborted and the console returns to the Position Busy mode. Any service change since the last Prime DN prompt (for initialize) or since the last successful datadump (for system reload) is lost and must be input again.

Feature interactions

Attendant Administration does not support the following features:

- Call Forward, Internal Calls
- Directory Number Delayed Ringing
- Message Registration
- Night Key for Direct Inward Dialing Digit Manipulation
- Period Pulse Metering
- Room Status
- Station Specific Authorization Code
- User Selectable Call Redirection

Attendant Consoles

It is not necessary to have the handset/headset plugged in while in the Program mode. Plugging in the handset/headset while in the Program mode has no effect.

Attendant Position Busy

If a console in the Attendant Administration mode is idle for more than 20 minutes, it automatically reverts to Position Busy. If the system is initialized or reloaded while the console is in Attendant Administration mode, Attendant Administration is aborted and the console is placed in Position Busy.

Attendant Supervisory Console

Attendant Administration mode can be entered directly from the supervisory console from Supervisory or Normal mode by pressing the program (PRG) key. The Supervisory mode does not need to be terminated first.

Automatic Wake Up

The Attendant Administration feature does not support data entry or changes for the Automatic Wake Up feature.

Call Forward No Answer/Flexible Call Forward No Answer

Attendant Administration can assign and change a Flexible Call Forward No Answer DN with the function key on the Attendant Console.

Call Hold, Deluxe

Deluxe Hold (DHLD) cannot be administered through the Attendant Administration feature.

Console Presentation Group Level Services

Attendants can dial the access code and activate the Administration mode. In this mode, they can modify the configuration of any telephone for this customer.

Controlled Class of Service, Enhanced

Attendant Administration cannot change Controlled Class Service restrictions (CCRS), ECC1 or ECC2, but can assign CLS keys to certain telephones.

Directory Number Delayed Ringing

The Attendant Administration feature is not supported.

End-to-End Signaling

While in the Attendant Administration mode, pressing the Attendant End-to-End Signaling key is ignored.

Hot Line

Use of an Attendant Console to change the database for Enhanced Hot Line is not supported.

ISDN Calling Line Identification Enhancements

Administration of a Calling Line Identification entry, for a set from an attendant console, is not supported.

Multiple Appearance Directory Number Redirection Prime

Multiple Appearance Directory Number Redirection Prime (MARP) TNs cannot be added, moved, or deleted with Attendant Administration. The DN information that displays on the console includes the MARP designation if applicable.

Attendant Administration activities, like changing key assignments or DN appearance, can change MARP TN assignments. If so, the CSC102 message appears on the teletype (TTY) indicating a new default MARP TN, as follows:

CSC102 DN nnnn NEW MARP l s c u (c u)

where:

nnnn = the DN associated with the MARP TN
l s c u (c u) = the new MARP TN assigned to DN nnnn (c u for Small Systems and Succession 1000 systems)

Multi-Party Operations

Attendant Administration allows certain station Classes of Service to be altered. The operation of Attendant Administration is modified so that if an attendant tries to alter either XFA or XFD Class of Service, then Three-party Service (TSA) Class of Service is disallowed. The TSA and XFA Classes of Service are mutually exclusive. When XFA is assigned, TSA will be disallowed if it was not configured. XFD is not mutually exclusive with TSA, but TSA will not be automatically assigned if the Class of Service is changed to XFD. TSA Class of Service cannot be assigned through Attendant Administration.

This feature can not be used to setup the Three-party Service TSA Class of Service.

Phantom Terminal Numbers (TNs)

The Attendant Administration feature does not support Phantom TNs. Phantom DNs cannot be configured on a non-phantom TN.

Remote Call Forward

Attendant Administration does not support the telephone programming associated with Remote Call Forward.

Speed Call, System

System Speed Call lists can be assigned using Attendant Administration.

Feature packaging

Attendant Administration (AA) package 54 requires Attendant Overflow Position (AOP) package 56.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Assign an Attendant Administration access code.

2 LD 12 – Add or change Attendant Administration key.

LD 15 – Assign an Attendant Administration access code.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	PWD	Customer related passwords
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ATAC	xxxx	New or changed Attendant Administration access code (maximum four digits). X preceding the access code removes it.
- - PWD2	xxxx	This password is programmed in LD 17 at the PWD2 prompt.

LD 12 – Add or change Attendant Administration key.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	ATT 1250 2250	Console type.
CUST	xx	Customer number, as defined in LD 15
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx PRG	Add an Attendant Administration key.

Feature operation

For details on feature operation, refer to the *Attendant Administration User Guide*.

Attendant Alternative Answering

Contents

This section contains information on the following topics:

Feature description 201
Operating parameters 204
Feature interactions 204
Feature packaging 206
Feature implementation 207
Feature operation 208

Feature description

Attendant Alternative Answering (AAA) allows customers to define a timing threshold for attendant calls. After the predefined time, the unanswered call presented to an idle loop key on an Attendant Console is forwarded to a predefined DN for alternate answering.

An unanswered call is forwarded to an idle or busy alternate DN. The call is subject to further call modification depending on the database configuration for the alternate DN.

When a call is presented to an idle loop key on the Attendant Console, the following occurs:

1 The system checks the attendant for AAA eligibility by checking for the AAA timer. The AAA timer activates the AAA feature.

2 When the timer expires, the unanswered call is forwarded to the Attendant Alternative Answering DN (AAA DN) defined for an individual attendant. Calls forwarded to the AAA DN are subject to the individual telephone's features, independent of the attendant. It is possible that the DN rung is not be the AAA DN.

3 After the alternate telephone has been reached, the Attendant Console releases the call.

4 If call termination is unsuccessful at the AAA DN, an error message is generated that explains the problem:

— If the error is because of an invalid AAA DN or tenant-to-tenant access denied condition, the call remains on the idle loop key for the attendant, and the AAA timer is not started again.

— For all other errors, the call remains on the attendant loop key and the AAA timer is restarted. The sequence is repeated until the call is answered at the console, disconnected by the caller, or terminated at the AAA DN.

When an Automatic Wake Up (AWU) recall is presented to the AWU key on the Attendant Console, the following occurs:

1 The AWU key buzzes, and the associated indicator fast flashes.

2 The attendant presses the AWU key to accept the recall.

3 The attendant presses the RLS key to release the call. An AWU recall must be acknowledged before any other calls can be presented to the attendant.

4 With AAA, the AWU call is presented to the attendant for the duration of the AAA timer. If an AWU recall is not acknowledged before the timer threshold, the recall is returned to the attendant queue to be presented later. The AWU recall will not be forwarded to the AAA DN.

If the AAA DN does not answer, call treatment is defined by the features allowed for the originally dialed DN. If the originally dialed DN is the attendant, call treatment is defined by the features allowed for the AAA DN.

The order listed below reflects the precedence when one or more call forwarding features is equipped:

1 Call Forward All Calls

2 Message Center

3 Call Forward No Answer

- — Flexible Call Forward No Answer
- — Second Level Call Forward No Answer
- — Call Forward by Call Type

4 Automatic Timed Recalls (slow answer)

For an unanswered call presented to a busy AAA DN, treatment is defined by the features enabled for that customer and the AAA DN telephone.

The order listed below reflects the precedence when one or more call forwarding features is equipped on the AAA DN:

1 Call Forward All Calls

2 Hunting

3 Call Waiting

4 Message Waiting (Direct Inward Dialing [DID] calls only) (if Message Waiting Forward Busy (MWFB) is enabled in LD 15)

5 Call Forward Busy (DID calls only)

If no Call Forwarding feature is defined for the busy AAA DN, the call remains on the Attendant Console, and the AAA timer is restarted. When the AAA timer expires, the call is again forwarded to the AAA DN.

Operating parameters

Attendant Alternative Answering (AAA) is defined and applicable on a customer basis only, not at the Console Presentation Group (CPG) level. AAA only handles calls presented to the console, not calls in the attendant queue. It is recommended that the AAA DN assigned to an attendant be within the same CPG as the attendant.

Only 63 Attendant Consoles can be assigned per customer. Only one AAA DN can be assigned per attendant; therefore, this feature is limited to 63 AAA DNs per customer, one for each Attendant Console.

With Night Service (NSVC) enabled and active, calls are rerouted to the Night Service DN. Calls presented to the NSVC DN are not subject to AAA.

The AAA DN must be a valid DN or ACD DN. If invalid, the call stays on the console.

The AAA DN defined is not subject to pretranslation. The AAA DN must be the actual DN.

This feature allows more than one backup of the attendant to be available, provided the designated alternative DN is defined as a member of a Call Pickup group or as a Multiple Appearance DN.

Feature interactions

Attendant Overflow Position

The Attendant Overflow Position (AOP) DN handles calls from the attendant queue if all Attendant Consoles are busy or in the Position Busy mode. Calls presented to the AOP DN are not subject to AAA.

Attendant Recall

Under Attendant Recall conditions (ARC), the initiator of the recall rings the destination side of the console, and the third party becomes the source. The AAA timer is applied to the source party. If the AAA timer expires, the destination is dropped, and the source is forwarded to the AAA DN. If the source party disconnects before the destination party, the AAA timer is restarted on the destination party still buzzing the attendant through the ARC key. The AAA timer is dropped if both parties disconnect.

Call Forward All Calls

Call Forward All Calls takes precedence over all other Call Forwarding features for a particular telephone. Calls forwarded by AAA are subject to the Call Forwarding conditions on the AAA DN.

Call Forward Busy

If Call Forward Busy is allowed for the AAA DN (and that DN is busy), a DID call is returned to the attendant and can again be eligible for AAA timing and operation.

Call Forward by Call Type

If Call Forward by Call Type is enabled on the AAA DN, calls are forwarded based on the Call Type of the originator.

Call Forward No Answer

When the AAA DN does not answer, the call can be forwarded by Call Forward No Answer (CFNA) to the DN defined as the CFNA DN for the originally dialed DN. If the originally dialed DN is the attendant, the call is forwarded to the CFNA DN defined for the AAA DN.

Call Pickup

The AAA DN can be assigned to a Call Pickup group to allow members of the same group to answer the call.

Centralized Attendant Service

The AAA timer is not applied to Centralized Attendant Service (CAS) calls routed from the remote CAS location through the Release Link Trunk to the main CAS attendant. All other internal or trunk calls presented to the CAS attendant at the main location are timed by AAA as usual.

If the remote CAS attendant presses the CAS key while a call is being presented, the presented call is subject to AAA timing and is forwarded to the AAA DN at the remote location after the timer expires.

Do Not Disturb

A DN in the Do Not Disturb (DND) mode is free to originate calls but appears busy to incoming calls. Call Forward All Calls takes precedence over DND indication on AAA DNs.

Group Hunt

A Pilot DN can be defined as an alternative DN. Calls forwarded to a Pilot DN as an alternative DN are directed to the next DN in the group.

Hunting

Calls directed to a busy AAA DN with Hunt defined are routed down the Hunt chain as defined for the AAA DN.

A Pilot DN for a hunting group can be defined as an AAA DN. Calls forwarded to a Pilot DN are directed to the next DN in the group.

Manual Line Service

When Attendant Alternative Answering (AAA) is defined, Manual Line Service follows the AAA parameters.

Message Center

If the AAA DN is a Message Center (MWC), then a Message Center call to the attendant and forwarded by AAA is still treated like a Message Center call.

Multi-Tenant Service

Tenant-to-tenant access must be allowed between an internal caller and the AAA DN. If caller-to-AAA access is denied, the call remains on the console until the call is answered or dropped.

Feature packaging

Attendant Alternative Answering (AAA) package 174 has no feature package dependencies; however, this package is mutually exclusive with Attendant Forward No Answer (AFNA) package 134.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the Attendant Alternate Answering feature.

2 LD 12 – Define the AAA DN for each Attendant Console affected.

LD 15 – Configure the Attendant Alternate Answering feature.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console Option.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ATIM	(0)-126	AAA timer in two-second increments. Odd numbers are rounded down. ATIM = 0 disables the feature

LD 12 – Define the AAA DN for each Attendant Console affected.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
AADN	xxx...x	Attendant Alternative Answering DN.

Feature operation

No specific operating procedures are required to use this feature.

Attendant and Network-Wide Remote Call Forward

This modification to the Remote Call Forward (RCFW) feature allows a user to program a Call Forward Directory Number from any attendant console or station throughout the network. A new RFW key on the attendant console allows an attendant to view any station's Call Forward status and to activate or deactivate Call Forward for a station.

Attendant Announcement

Contents

This section contains information on the following topics:

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Feature description

The Attendant Announcement (AANN) feature provides announcements for calls terminating on the attendant, attendant queue or night station. Announcements continue to play until the attendant answers the call.

Attendant Announcement is enabled on a route basis at the Attendant Announcement (ATAN) prompt.

An Attendant Announcement can be provided when a call from the Public Network terminates over an MCDN trunk to the attendant or night station. For this functionality, set the ATAN prompt to PSTN in LD 16. An announcement is provided when the incoming call over the network is marked as a PSTN call. Network Attendant Services (NAS) must be enabled for the TIE trunk's D-channel.

Attendant Announcement types

The Attendant Announcement feature provides different announcements based on the state of the call.

Configure the following announcement types in LD 56:

- Announcement when terminating to the Attendant (ANAT)

  When a call is dialed directly, or intercepted, to the attendant, the caller receives an ANAT announcement.

- Announcement when Night Service is activated (ANNS)

  When a call is terminated to the night station or the night service queue, the caller receives an ANNS announcement.

- Announcement when Call Forward No Answer to the Attendant (ANFA)

  When a Call Forward No Answer (CFNA) call is redirected to the attendant, the caller receives an ANFA announcement.

- Announcement when Call Forward Busy to the Attendant (ANFB)

  When a Call Forward Busy (CFB) or Hunt call is redirected to the attendant, the caller receives an ANFB announcement.

- Announcement when Slow Answer Recall to the Attendant (ANSR)

  When a call is extended by the attendant and the call is not answered within Recall Timer (RTIM) time, the caller is redirected to the attendant and receives an ANSR announcement.

- Announcement on Attendant Extended Calls (ANXC)

  When an attendant transfers a trunk call to an extension, the caller receives an ANXC announcement until the requested party goes off hook.

- Announcement when Overflowed or Forwarded (ANOF)

  If a customer uses the Attendant Overflow Position (AOP) or Attendant Alternative Answer (AAA) features, the call is redirected after a specific time to a predefined telephone. The caller receives an ANOF announcement until the call is answered.

Table 6 summarizes the types of announcements provided to the caller when a call is terminated to the attendant, attendant queue, night station or night queue.

Table 6
Announcement received at termination

Type of call	Call destination	Announcement received
Direct calls or abandoned calls	Attendant or attendant queue	ANAT
Direct calls or abandoned calls	Night station or night queue	ANNS
Call Forward No Answer treatment (CFNA)	Attendant, attendant queue, or night station	ANFA
Call Forward Busy treatment (CFB)	Attendant, attendant queue, or night station	ANFB
Slow Answer Recall	Attendant, attendant queue, or night station	ANSR
Attendant Extended calls	Attendant, attendant queue, or night station	ANXC
Overflowed or Forwarded calls	Attendant, attendant queue, or night station	ANOF

Special options

During normal operations, when a call terminates to the night station, the ANNS announcement is given. This also applies to redirected calls. However, on small systems where the switch is in permanent night mode, call redirection information cannot be used. In this case, Call Forward No Answer calls terminate to the night station and the caller receives an ANNS announcement. If the Night station announcement priority (NIPR) option in the announcement profile is set to “NO”, calls redirected to the night station receive an appropriate greeting.

If an announcement is required only when the call is in the attendant or night service queue, set the Attendant Queue (ANQU) option to “YES”.

Announcement source types

Either of the following external devices provides attendant announcements to the caller:

- Tone interface
- Recorded Announcement (RAN) trunk interface (for example, MIRAN)

Tone interface announcements

When announcements are provided through the tone interface, they are treated as tones.

Tone announcements require a digital speech generator connected to the faceplate connector of the Extended Conference (XCT) card or Tone and Digit Switch (TDS) card.

Tone interface announcements are configured in LD 56.

Tone interface announcements play from the beginning of the announcement until the attendant answers the call. No initial greeting can be played.

RAN trunk interface announcements

Attendant announcements can be provided by existing RAN trunks.

To ensure that callers hear the announcements from the beginning, configure the Recorded Announcement with a Delay Dial (DDL) at the Start Arrangement (STRT) prompt in LD 16.

Answer Supervision for RAN trunks

Use the Answer Supervision (ASUP) prompt in LD 16 to return Answer Supervision by RAN to the originator.

Post-RAN post treatment

Existing RAN functionality allows an announcement to repeat up to fifteen times. Post-RAN treatment is followed after the defined number of repetitions. The number of repetitions and Post-RAN treatment are defined in LD 16 at the REP and POST prompts respectively.

For the Attendant Announcement feature, Post-RAN treatment uses RAN Hunting. RAN Hunting allows a new RAN trunk to be connected after the preceding RAN trunk is terminated. This allows a general Recorded Announcement to play once. When this announcement finishes, it then switches to another announcement.

If RAN Hunting is configured to connect to the same route, Hunting does not occur. Therefore, the same Recorded Announcement repeats in a continuous loop. If RAN Hunting is not configured, the current RAN route is used.

Alternative Attendant Announcement treatment

With Alternative Attendant Announcement (AAT) treatment, different announcements are provided to the caller depending on the time and date. For example, a “Good morning!” greeting can be played until noon and then the greeting is switched automatically to “Good afternoon!”.

When you enable AAT in LD 16, you have the following options:

- Alternative Attendant Announcement Time of Day (AATO)
- Alternative Attendant Announcement Day of Week (ADAY)
- Alternative Attendant Announcement Holiday (AHOL)

You can configure up to four different optional times of day and four different optional days of week. Configure these options to select an Alternative Announcement Table (AATB). Only one alternate time and announcement table can be used in the Route Data Block.

If a caller calls within a period specified by one of the Alternative Attendant Announcement options, the Alternative Announcement Table is used.

If the Alternative Attendant Announcement treatment is used with Call Redirection by Time of Day or Call Redirection by Day of Week, the four alternative options must be shared between the two features.

If MIRAN is used as a RAN source, the Alternative Attendant Announcement option can be disabled, as MIRAN uses this capability. This helps to reduce the number of RAN ports.

Attendant Alternative Answer

When a call is originated by a trunk, it must be answered in order for the Attendant Announcement to be provided. When Call Answer functionality is activated, the call registers as an answered call.

For Call Answer functionality, you must select one of the following options at the Attendant Alternative Answer Option (AAAO) prompt in LD 16:

- No Call Answer (NO)
- Call Answer on Announcement (CAA)
- Call Answer Forced (CAF)

No Call Answer

No Call Answer is the default operation. With this option, No Call Answer is provided by this feature.

Select this option for trunks where it is not necessary to answer the trunk in order to open the speechpath.

Call Answer on Announcement

When you select this option, a connect message is sent to the originating trunk only when an announcement is provided. An answer is not provided if the incoming call does not terminate to an attendant.

An answer is provided in the following cases:

- a call terminates to the attendant, attendant queue or night station.
- the Call Answer option is enabled
- an external announcement has been configured in LD 56.

Call Answer Forced

Only select this option for cases when tone announcement will be used as the announcement source and an announcement is necessary for all calls terminating to the switch.

When this option is activated, an error message is displayed to indicate that all calls are answered immediately.

With Call Answer Forced, an answer is provided in the following cases:

- a call terminates to the attendant, attendant queue or night station.
- the call answer option is enabled
- an external announcement has been configured in LD 56.

Operating parameters

If a greeting is not defined for one of the announcement types, the caller receives a normal ringback tone. This generates an error message to the maintenance terminal.

After system initialization, calls receiving an announcement are not restored. The calls are dropped and the caller hears silence.

If a caller calls the night station directly, no attendant announcement is provided.

Attendant Announcement is not provided on series calls.

Feature interactions

Attendant Alternative Answering

If the call to the attendant receives an attendant announcement and the call is forwarded to the Attendant Alternative Answering DN, the announcement is removed and an ANOF announcement is provided, if configured.

Attendant Barge-In

A busy tone is provided to the attendant when the operator barges into a trunk that is receiving an attendant announcement.

Attendant Clearing during Night Service

When the attendant goes into night service and a call is in the attendant queue, the call is routed to the night DN and receives the appropriate announcement defined for the night station.

Attendant Forward No Answer

If a call is presented to the attendant, the call receives an announcement, and is forwarded to the night station, the call is requeued. If the call goes to the night station, the caller hears an ANNS announcement, if configured.

Attendant Interpositional Transfer

When an incoming call with Attendant Announcement enabled is transferred to another attendant, no announcement is provided.

Attendant Overflow Position

If a call is presented to the attendant while receiving an announcement and the call is then forwarded to the Attendant Overflow Position DN, the announcement is not removed. The ANOF announcement is provided, if configured.

Attendant Recall

Attendant Announcement does not support the Attendant Recall feature.

Automatic Call Distribution

Automatic Call Distribution (ACD) applies when the night DN is an ACD DN. No announcement is provided when a call terminates to the ACD queue. ACD announcements must be configured instead.

Automatic Timed Reminders

An Automatic Timed Reminders recall receives the appropriate announcements.

Call Detail Recording Time to Answer

Attendant Announcement does not affect Call Detail Recording Time to Answer. A separate CDR for the RAN trunk is generated by the RAN answered calls.

Call Forward All Calls

An Attendant Announcement is provided if the night station activates Call Forward All Calls (CFAC).

An Attendant Announcement is provided when the call terminates to the attendant.

Call Forward No Answer
Call Forward Busy
Slow Answer Recall

Call Forward No Answer (CFNA), Call Forward Busy (CFB) or Slow Answer Recall announcements take precedence over direct calls to the attendant, attendant queue or night station. Announcement when terminating to the Attendant (ANAT) or Announcement when Night Service is active (ANNS) is the standard announcement provided for other calls. Ringback tone is provided to the caller if an announcement is not defined.

Call Redirection by Time of Day
Call Redirection by Day of Week

For Call Redirection by Time of Day and Call Redirection by Day of Week, it is possible to configure up to four options. If Attendant Announcement is configured to use either Call Redirection by Time of Day or Call Redirection by Day of Week, three options remain.

Centralized Attendant Service

Centralized Attendant Service does not support Attendant Announcement.

DPNSS1

The Attendant Announcement feature does not support DPNSS-originated calls.

Direct Inward Dialing Call Forward No Answer Timer

DID Forward No Answer (DFNR) calls receive the Call Forward No Answer announcement when it is terminated to the attendant.

EuroISDN Connected Number

If a call is presented to the attendant, an attendant announcement is provided to the caller. The dialed DN is provided as a connected number.

MCDN-QSIG Gateway

The MCDN-QSIG Gateway is not affected by the Attendant Announcement feature. Attendant Announcement uses existing Network Attendant Services (NAS) information to determine whether an announcement should be given.

Trunk Anti-Tromboning

Trunk-to-trunk connections are optimized when they receive ANSWER treatment. Attendant Announcement answers a trunk call; however, the actual call is not established. The trunk is in an answer state, but it is still present in the attendant queue.

Trunk Anti-Tromboning (TAT) is not triggered during Attendant Announcement. TAT is triggered to optimize the call when the console answers the call.

Recorded Overflow Announcement

Attendant Announcement takes precedence over Recorded Overflow Announcement.

Slow Answer Recall

Slow Answer Recall calls receive ANSR announcement when specified.

Virtual Network Service

Announcements are not provided on internal VNS calls. If ATAN is set to "YES", no VNS calls receive Attendant Announcement.

Feature packaging

The Attendant Announcement feature introduces Attendant Announcement (AANN) package 384.

This feature also requires the following existing packages:

- Recorded Announcement (RAN) package 7 (if RAN Announcements are used)
- Attendant Overflow Position (AOP) package 56 (if AOP is used)
- Flexible Tones and Cadences (FTC) package 125

- Attendant Forward No Answer (AFNA) package 134 (if AFNA is used)
- Network Attendant Service (NAS) package 159 (if used over MCDN network)
- Message Intercept (MINT) package 163
- Attendant Alternative Answering (AAA) package 174 (if AAA is used)
- Recorded Announcement Broadcast (RANBRD) package 327 (if the broadcast facility of the RAN trunk is used)

Feature implementation

Task summary list

Use the following to configure announcements provided by XCT/TDS tone service:

1 LD 56 – Configure the Attendant Announcement table.

2 LD 56 – Configure tone announcement for Small Systems and Succession 1000 systems.

3 LD 56 – Configure tone announcement for Large Systems.

4 LD 16 – Enable the Attendant Announcement.

Use the following to configure announcements provided by RAN services:

1 LD 16 – Configure RAN routes for Attendant Announcement.

2 LD 56 – Configure the Attendant Announcement table for RAN usage.

3 LD 16 – Configure Route Data Block for Attendant Announcement.

Announcements provided by XCT/TDS tone service

LD 56 – Configure the Attendant Announcement table.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AANN	Attendant Announcement data block.
CUST	xx	Customer number, as defined in LD 15
TBL	0-31	Announcement table number.
- NIPR		Nightstation Announcement Priority.
	(NO) YES	ANNS is not provided on each call to the night station ANNS is provided on each call to the night station.
- ANQU	(NO) YES	Announcement is not provided on calls in the attendant queue or night service queue only. Announcement is provided on calls in the attendant queue or night service queue only.
- ANAT	aaa	Announcement when terminating to the Attendant, where: aaa= SRC1 - SRC8 source entry of the appropriate tone table.
- ANNS	aaa	Announcement when terminating to night station, where: aaa = SRC1 - SRC8 source entry of the appropriate tone table.
- ANFA	aaa	Announcement when Call Forward No Answer to Attendant, where: aaa = SRC1 - SRC8 source entry of the appropriate tone table.
- ANFB	aaa	Announcement when Call Forward Busy to Attendant, where: aaa = SRC1 - SRC8 source entry of the appropriate tone table.
- ANSR	aaa	Announcement when Slow Answer Recall, where: aaa = SRC1 - SRC8 source entry of the appropriate tone table.

Prompt	Response	Description
- ANXC	aaa	Announcement on Attendant Extended Calls, where: aaa = SRC1 - SRC8 source entry of the appropriate tone table.
- ANOF	aaa	Announcement on Attendant Overflow Calls, where: aaa = SRC1 - SRC8 source entry of the appropriate tone table.

LD 56 – Configure tone announcement for Small Systems and Succession 1000 systems.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FTC	Flexible Tones and Cadences.
TABL	0-31	Define tone table number.
...		
SRC	YES	Source.
SRC1 - XTON - XCAD	 (0)-255 (0)-255	Source that indicates announcement channel of the hardware. XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.
SRC2 - XTON - XCAD	 (0)-255 (0)-255	Source that indicates announcement channel of the hardware. XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.
- SRC3 - XTON - XCAD	 (0)-255 (0)-255	Source that indicates announcement channel of the hardware. XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.
SRC4 - XTON - XCAD	 (0)-255 (0)-255	Source that indicates announcement channel of the hardware. XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.

LD 56 – Configure tone announcement for Large Systems.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change new data.
TYPE	FTC	Flexible tone and cadences.
TABL	0-31	Define tone table number.
...		
SRC	YES	Source.
SRC1		Source that indicates announcement channel of the hardware, where: xx = (0)-255.
- TDSH - XTON - XCAD	1 0 0 xx xx yy	TDS Hex XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.
SRC2		Source that indicates announcement channel of the hardware, where: xx = (0)-255.
- TDSH - XTON - XCAD	1 0 0 xx xx yy	TDS Hex XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.
SRC3		Source that indicates announcement channel of the hardware, where: xx = (0)-255.
- TDSH - XTON - XCAD	1 0 0 xx xx yy	TDS Hex XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.
SRC4		Source that indicates announcement channel of the hardware, where: xx = (0)-255.
- TDSH - XTON - XCAD	1 0 0 xx xx yy	TDS Hex XCT (NT8D17 Conference/TDS) Tone code. XCT (NT8D17 Conference/TDS) Cadence number.

LD 16 – Enable the Attendant Announcement.

Prompt	Response	Description
REQ	NEW CHG	Add a new data. Change existing data.
TYPE	RDB	Route Data Block.
...		
TKTP		Trunk Type
	a..a	Attendant Announcement is available on DID, TIE and COT trunks only.
...		
ATAN		Attendant Announcement.
	(NO) YES PSTN	No Attendant Announcement. Enable Attendant Announcement on this route. Enable Attendant Announcement on this route for PSTN calls only (for MCDN trunks only).
- ATBL	xx	Announcement profile table, where: xx = 0-31 This number should correspond with what you set at the AANN prompt to in LD 56.
- AAT		Alternative Attendant Announcement.
	(NO) YES	Disable Alternative Attendant Announcement. Enable Alternative Attendant Announcement.
--AATO	(0) - 3	Alternative Attendant Announcement Time of Day option.
--ADAY	(0) - 3	Alternative Attendant Announcement Day of Week option.
--AHOL	(0) - 3	Alternative Attendant Announcement Holiday option.
--AATB	xx	Announcement Profile Table for Alternative Announcement, where: xx = 0-31 This number should correspond with what you set at the AANN prompt to in LD 56.

Prompt	Response	Description
- AAAO		Attendant Alternative Answer Option.
		This option is for Tone Announcements only.
	(NO) CAA CAF	No call answer is given Call answer will be given on announcement. Call answer will be given forced.

Announcements provided by RAN services

LD 16 – Configure RAN routes for Attendant Announcement.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
...		
TKTP		Trunk Type.
	RAN	RAN route.
RTYP	MCON	Continuous multichannel.
REP	1-15	Number of repetitions of this RAN.
STRT	DDL	Delay call connection until start of announcement.
BDCT		Broadcast Capability.
	(NO) YES	Deny RAN Broadcast Capability for this route. Allows RAN Broadcast Capability for this route.
WAIT	RGB	Provide ringback tone for calls queuing for RAN trunk.

Prompt	Response	Description
ASUP		Answer Supervision.
	(NO)	Answer Supervision is controlled in the RDB of the incoming trunk route.
	YES	Return Answer Supervision.
RANH	0-511	RAN route number when Attendant Announcement is completed.

LD 56 – Configure the Attendant Announcement table for RAN usage.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AANN	Attendant Announcement data block.
CUST	xx	Customer number, as defined in LD 15
TBL	0-31	Announcement table number.
- NIPR		Night station Announcement Priority.
	(NO) YES	ANNS is not provided on each call to the night station. ANNS is provided on each call to the night station.
- ANQU		Attendant Queue.
	(NO)	Announcement is not provided on calls in the attendant queue or night service queue only.
	YES	Announcement is provided on calls in the attendant queue or night service queue only.
- ANAT	aaa	Announcement when terminating to the Attendant, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.

Prompt	Response	Description
- ANNS	aaa	Announcement when terminating to night station, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.
- ANFA	aaa	Announcement when Call Forward No Answer to Attendant, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.
- ANFB	aaa	Announcement when Call Forward Busy to Attendant, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.
- ANSR	aaa	Announcement when Slow Answer Recall, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.
- ANXC	aaa	Announcement on Attendant Extended Calls, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.
- ANOF	aaa	Announcement on Attendant Overflow Calls, where: aaa = R000 - R511 announcement is provided through the RAN announcement for large system. aaa = R000 - R128 announcement is provided through the RAN announcement for small system.

LD 16 – Configure Route Data Block for Attendant Announcement.

Prompt	Response	Description
REQ	NEW CHG	Add a new data. Change existing data.
TYPE	RDB	Route Data Block.
...		
TKTP		Trunk Type.
	a..a	Attendant announcement is available on DID, TIE and COT trunks only.
...		
ATAN		Attendant Announcement.
	(NO) YES PSTN	No Attendant Announcement. Enable Attendant Announcement on this route. Enable Attendant Announcement on this route on PSTN calls only (For MCDN trunks only).
- ATBL	xx	Announcement profile table, where: xx = 0-31 This number should correspond with what you set the AANN prompt to in LD 56.
- AAT		Alternative Attendant Announcement.
	(NO) YES	Disable Alternative Attendant Announcement. Enable Alternative Attendant Announcement.
--AATO	(0) - 3	Alternative Attendant Announcement Time of Day option.
--ADAY	(0) - 3	Alternative Attendant Announcement Day of Week option.
--AHOL	(0) - 3	Alternative Attendant Announcement Holiday option.
--AATB	xx	Announcement Profile Table for Alternative Announcement, where: xx = 0-31 This number should correspond with what you set at the AANN prompt to in LD 56.
- AAAO	(NO)	No Call Answer is given (for Tone Announcement only).

Feature operation

No specific operating procedures are required to use this feature.

Attendant Barge-In

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Feature description

Attendant Barge-In allows the attendant to establish a connection with any trunk in the system to verify that the trunk is in working order. When Barge-In is active, a 256 millisecond burst of tone is sent to the connected parties every six seconds to indicate the presence of the attendant.

Operating parameters

Barge-In can only be used for trunks with Warning Tone Allowed (WTA) Class of Service. All parties connected to the trunk when the attendant attempts to barge in must have WTA Class of Service.

If equipped, the Barge-In key must be assigned to key 1 of the console flexible feature strip.

The system must be equipped with a conference loop.

Feature interactions

Automatic Redial

Attendant Barge In is not allowed to a trunk that is currently used for the Automatic Redial call redialing. This is done to avoid creating a conference when the tone detector is involved.

Call Forward/Hunt Override Via Flexible Feature Code

Using Call Forward/Hunt Override Via FFC after activation of Barge-in, Busy Verify or Break-in is not allowed. Attempts will be canceled and overflow tone will be returned.

Using post-dial Break-in after dialing the Call Forward/Hunt Override FFC is possible after encountering a busy set, if Break-in is enabled.

Call Page Network Wide

For external Call Page Network Wide (PAGENET) uncontrolled calls, Attendant Barge-In is blocked at the Paging node, per existing operation. For external PAGENET controlled calls, Attendant Barge In is blocked at both the originating and Paging node.

Charge Account and Calling Party Name

A charge account number cannot be entered when Attendant Barge-In or Attendant Busy Verify is active. Barge-In cannot be used to connect to a trunk after an account number has been entered.

China – Attendant Monitor

When China (CHINA) package 285 is equipped, the normal operation of Barge-In changes slightly. The repeatable tone can be configured with the (TOA)/TOD option.

If an attendant is monitoring a trunk, a second attendant defined at the same customer location is blocked from Barging In to any trunk involved in the monitored call.

If an attendant is Barged-In with a trunk, a second attendant defined at the same customer location will be blocked from monitoring any party involved in the monitored call.

Conference

Conference Control cannot be activated if an attendant has used Barge-In or during a conference that involves a trunk.

End-to-End Signaling

While in the Attendant Barge-In mode, the console cannot enter Attendant End-to-End Signaling mode.

Intercept Computer Dial from Directory - Pre-dial Operations

It is possible for an attendant to Barge-in, in the following manner:

- Press an idle loop key, and press the Barge-in key from the attendant console.
- Dial a Route Access code and Route member from the ICT (which must be configured in such a way that it is possible to dial the Route access code and Route member from the dialing key).

ISDN Semi Permanent Connections for Australia

When an attendant attempts to Barge-In on 2.0 Mbps Primary Rate Interface B-channel used as an ISPC link with the Central Office, a fast busy tone is provided.

Uninterrupted Line Connections

Attendant Barge-In cannot be applied to stations with a Warning Tone Denied Class of Service.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 12 – Add or change a Barge-In key on Attendant Consoles..

2 LD 10 – Allow or deny a warning tone Class of Service for analog (500/2500 type) telephones.

3 LD 11 – Allow or deny a warning tone Class of Service for proprietary telephones..

4 LD 14 – Allow or deny warning tone Class of Service for trunks.

LD 12 – Add or change a Barge-In key on Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	1 BIN	Add a Barge-In key.

LD 10 – Allow or deny a warning tone Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	(Allow) deny warning tone.

LD 11 – Allow or deny a warning tone Class of Service for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	(Allow) deny warning tone.

LD 14 – Allow or deny warning tone Class of Service for trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	COT DID FEX RAN TIE WATS	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	(Allow) deny warning tone.

Feature operation

To establish a connection on a trunk, follow these steps:

1 Select an idle loop key.

2 Press **Barge-In**.

3 Dial the route access code and the trunk member number, followed by the octothorpe (#).

The possible results are the following:

- dial tone (trunk is idle and working)
- conversation (trunk is busy and working)
- modem carrier tone (long distance trunk is working)
- fast busy (trunk is either disabled or has Warning Tone Denied Class of Service)

If you hear fast busy, check the trunk again before reporting a problem.

Attendant Break-In

Contents

This section contains information on the following topics:

Feature description 237
Operating parameters 238
Feature interactions 239
Feature packaging 244
Feature implementation 245
Feature operation 245

Feature description

The Attendant Break-In (BKI) feature simplifies the process required if an attendant must break in to an established call. When an attendant receives an urgent call and dials the destination DN, that DN may be busy. The attendant may then have to break in to the call. This feature provides a new key on the attendant console: the Break-In key. This feature allows the attendant to extend a call to a busy extension through a simple key operation.

The break-in process involves the following steps:

1 Use the Break-In key upon receiving the busy tone. This action establishes a conference between the attendant and the members of the established call (but excludes the incoming call). Parties hear the intrusion tone; secrecy is preserved.

2 Announce that an important call is waiting and request that the other parties disconnect from the call.

3 Extend the incoming call to the previously busy DN as soon as the other parties disconnect.

By using the Break-In key before dialing the destination DN, the attendant can override features such as Call Forward and Hunting.

Operating parameters

The Attendant Break-In feature is supported on analog (500/2500 type) telephones and proprietary telephones.

A console can have only one Break-In key.

A break-in connection cannot be put on hold.

Only one attendant at a time can break in to a call.

Attendant Break-In does not operate in the following situations:

- A party to the established call has Override Denied or Warning Tone Denied Class of Service
- The established call involves any of the following:
  - An attendant
  - Multi-frequency Compelled (MFC) device type
  - Digitone Receiver (DTR) device type
  - Page trunk
  - Dictation trunk
  - Recorded Announcement trunk

— Integrated Voice and Message System (IVMS)

- The destination DN is on an outgoing trunk call. If the station is involved with an outgoing trunk call, the call is established when End of Dialing (EOD) times out, the number is dialed, or the trunk is answered.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN and the source side Predial Break-in features are mutually exclusive for the same call. If the SACP key lamp is lit when the Break-in key is pressed to start a Predial Break-in attempt, the Break-in key is ignored. On the contrary, if the Break-in key lamp is lit and no call attempt is made on the source side when the SACP key is pressed to start an Attendant Blocking of DN, the SACP key is ignored.

If a Break-in attempt is made for an Attendant Blocking of DN call, the Break-in attempt will be considered to be temporarily denied.

It will be possible to Break-in on the destination side with an Attendant Blocking of DN call on the source side of the Attendant Console. The same limitations to Break-in will apply as if the source side call is a normal call.

Attendant Break-In to Inquiry Calls

All other interactions are the same as for the Attendant Break-In feature.

Attendant Busy Verify

The attendant can use the Break-In key instead of Busy Verify to break in to an established call. Attendant Break-In simplifies this process.

Automatic Call Distribution

Once the destination DN has established the call with the Automatic Call Distribution (ACD) agent, the attendant can break in to the call. If the destination DN is in the ACD queue, Attendant Break-In is temporarily denied.

Automatic Redial

Attendant Break-In and Attendant Busy Verify are not permitted on a proprietary set that is used for an Automatic Redial (ARDL) call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Busy Verify on Calling Party Control Call

Local Attendant Break-In will be temporarily denied if the desired party is already in a toll operator Break-In conference or on a Special Service call, or awaiting the Special Operator signal. Local attendant/toll operator Break-In will be temporarily denied if the desired party is established on an incoming toll call.

Call Forward All Calls

By pressing the Break-In key before dialing the destination DN, the attendant can override call forwarding on the destination DN. The attendant may not apply Camp-On to a telephone with Call Forward active.

Call Forward/Hunt Override Via Flexible Feature Code

The use of Call Forward/Hunt Override Via FFC after activation of Barge-in, Busy Verify or Break-in is not allowed. Attempts will be canceled and overflow tone will be returned.

The use of post-dial Break-in after dialing the Call Forward/Hunt Override FFC is possible after encountering a busy set, if Break-in is enabled.

Call Forward, Break-In and Hunt Internal and External Network Wide

If the Internal/External definition in LD 15 is set to YES, a call is treated as internal or external on a network wide basis.

Call Hold, Permanent
Call Park

The attendant cannot break in to a call on hold or a parked call.

Call Page Network Wide

For external Call Page Network Wide (PAGENET) uncontrolled calls, Attendant Barge-In is blocked at the Paging node, per existing operation. For external PAGENET controlled calls, Attendant Barge In is blocked at both the originating and Paging node.

Call Transfer

The attendant cannot break in to a call that is being transferred until the transferred call is connected.

Call Waiting
Camp-On

If the destination DN has a camped-on incoming trunk call, the attendant cannot extend the urgent incoming call as a Camp-On call.

Camp-on, Forced

Telephones with a toll operator break-in call cannot be camped on to. Overflow tone is returned to telephones attempting Forced Camp-on.

China – Attendant Monitor

If an attendant is monitoring a DN, a second attendant defined at the same customer site will be blocked from Breaking In to any party involved in the monitored call.

If an attendant is in a Break-In situation with a DN, a second attendant defined at the same customer site will be blocked from monitoring any party involved in the monitored call.

China Number 1 Signaling - Called Party Control

Attendant Break-In is not allowed on an outgoing Called Party Control call.

Conference

If the attendant cannot break in to a conference call because the call is supporting the maximum number of callers, busy tone continues and the Break-In key lamp flashes.

Digit Display

During Attendant Break-In, the Attendant Console Digit Display shows the DN of the incoming call and the destination DN until the attendant extends the incoming call to the destination DN and releases the connection.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Group Hunt

Attendant Break-in will not be supported when dialing a Pilot DN directly.

Hold

The attendant cannot break in to a call on hold.

Hunting

If the destination DN is in a Hunting chain with some idle DNs, the Break-In request goes to the first idle DN in the chain. To prevent this occurrence, the attendant can press the Break-In key prior to dialing the destination DN.

Intercept Computer Dial from Directory - Post-dial Operation

Attendant Break-in

An attendant can break-in to a call by:

- Dialing an extension DN from the Intercept Computer.

Pressing the Break-in key on the Attendant Console.

Make Set Busy
Do Not Disturb

For a telephone with Make Set Busy or Do No Disturb in effect, Break-In is temporarily denied to the attendant. The Break-In lamp uses slow flash to indicate this situation. Using the Break-In key prior to dialing the destination DN circumvents this situation. After the Break-In, the telephone returns to its prior status.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Make Set Busy that may be applied to the set.

Meridian 911 Call Abandon

Since an abandoned call does not have a speech path established, the Break-In deny treatment is given to the attendant so that Break-In cannot occur.

Multiple Appearance Directory Number Redirection Prime

The attendant may get a busy tone if all the telephones with the required DN are busy. Break-In permits the attendant to break in to the connection with the least restricted TN. Where more than one TN exists that meets this criterion, Break-In chooses the one at the bottom of the DN block.

Multi-Party Operations – Three-Party Service

Break-In is not allowed to the party receiving the patience tone or the misoperation ringback.

Multi-Party Operations Enhancements

Attendant Break-in is not allowed to a connection in which a party is receiving Patience Tone or recall of misoperation ringback.

On Hold on Loudspeaker

It will not be possible to Break-in into a call on loudspeaker as it is effectively on hold at the set.

Override

When one telephone has overridden an existing call to establish a conference call, Break-In is temporarily denied. The attendant is notified by the override tone.

Priority Override

Telephones with a toll operator break-in call cannot be overridden. Overflow tone is returned to telephones attempting Priority Override.

Override, Enhanced

Telephones with a toll operator break-in call cannot be camped on to or overridden. Overflow tone is returned to telephones attempting either Forced Camp-on or Priority Override.

Periodic Camp-on Tone

The Periodic Camp-On Tone has precedence over Break-In intrusion tone.

Semi-Automatic Camp-On

The attendant can Break-In to an established call and apply Semi-automatic Camp-On to the desired party. The attendant may press the SACP key before or after the Break-In.

Source Included when Attendant Dials

The operation of the Break-In feature is not affected, except that the source receives busy tone before the attendant presses the Break-In (BKI) key.

Trunk Barring

Trunk Barring does not result in intercept treatment for Toll Operator Break-In.

Feature packaging

Attendant Break-In (BKI) is package 127.

Feature implementation

LD 12 – Assign the Break-In key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Attendant Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx BKI	Break-In key.

Feature operation

The operator can press the Break-In key either before or after dialing the destination DN. Break-In operates slightly differently in these two situations, as described below.

Post-Dial Break-In

For post-dial break-in

1 The attendant answers an incoming external call.

2 The attendant dials the destination DN.

3 The attendant receives the busy tone (unless the destination DN allows Camp-On or Call Waiting).

4 The attendant presses the Break-In key.

5 If allowed, the attendant joins the call on the destination DN to announce the incoming call and request that other parties disconnect. (See Table 7 on page 246 for an explanation of console break in states.)

6 After the other parties disconnect, the attendant extends the incoming call to the destination DN.

Pre-Dial Break-In

For pre-dial break-in

1 The attendant answers an incoming external call.

2 The attendant presses the Break-In key.

3 The attendant dials the destination DN.

4 If the destination DN is busy, the attendant hears the busy tone; processing is the same as for Post-Dial Break-In above.

If the destination DN is not busy, the DEST lamp flashes and the Break-In lamp goes dark. The attendant hears the ringback tone. Pressing the Break-In key a second time causes normal call processing for an idle line.

If the destination DN is invalid, the attendant hears the overflow tone and the Break-In lamp goes off. To return to the source call, the attendant presses the Release Destination key.

Table 7 describes the possible Attendant Console break in states. These states depend on several factors:

- whether the source call is an external call
- the type of call in effect at the destination DN
- the combination of features allowed at the destination DN
- whether the attendant pressed the Break-In key before or after dialing the destination DN

Table 7
Attendant Console break-in states (Part 1 of 2)

Console State	Lamp State	Description
ALLOW	Destination = LIT Break-In = LIT Tone = INTRUSION	The attendant can break in to the call and extend the incoming call.
CONSULT ONLY	Destination = FLASH Break-In = LIT Tone = INTRUSION	The attendant can break in to the call but cannot extend the incoming call.

Table 7
Attendant Console break-in states (Part 2 of 2)

Console State	Lamp State	Description
TEMPORARILY DENIED 1	Destination = FLASH Break-In = FLASH Tone = BUSY/ OVERRIDE	The attendant temporarily cannot break in to the call, and may attempt the break in later.
TEMPORARILY DENIED 2	Destination = FAST FLASH Break-In = FLASH Tone = OVERFLOW	The attendant temporarily cannot break in to the call.
DENIED	Destination = FLASH Break-In = DARK Tone = OVERFLOW	The attendant cannot break in to the established call or extend the incoming call.
BREAK-IN IGNORED	Destination = FLASH Break-In = DARK Tone = RING BACK	The attendant cannot break in. The attendant should make a second break in attempt.
INVALID DN	Destination = FLASH Break-In = DARK Tone = OVERFLOW	The attendant attempted to reach an invalid DN. The attendant should dial the correct destination DN.

Attendant Break-In Busy Indication and Prevention

Contents

This section contains information on the following topics:

Feature description 249
Operating parameters 250
Feature interactions 251
Feature packaging 251
Feature implementation 251
Feature operation 251

Feature description

This feature, operating either in a standalone or Integrated Services Digital Network (ISDN) environment, provides enhancements to the Attendant Break-in feature. This feature is described more fully in *ISDN Primary Rate Interface: Features* (553-3001-369).

Break-in Busy Indication

If an attendant, during a break-in operation, dials a busy extension, the Attendant Console display provides one of the following customer-defined indications:

- three dashes, appended to the end of a digit display (if the busy station is involved in an external call)
- a mode digit, appended to the end of a digit display

In a non-ISDN environment, the mode digit indicates one of the states:

1 = Station is busy on an external call, or station is busy on an off-net call.

2 = Station is busy on an internal call, or station is busy on an on-net call.

3 = Station is busy on a non-established call; for instance, dialing, ringing, or announcement. Or, station is busy on a conference call.

4 = Station is in line lockout.

In an ISDN Primary Rate Interface (PRI) environment, the mode digit indicates one of the following states:

1 = Station is busy on an off-net call, or involved in a conference call.

2 = Station is busy with on-net call, and is not involved in a conference call.

3 = Station is busy on a non-established call; for instance, dialing, ringing, or announcement.

4 = Station is in line lockout.

Break-in Prevention

A Break-in to External Call Denied (BIXD) option is provided to the customer which, if selected, temporarily denies Break-in to a party involved in an external call. This applies to both pre-dial and post-dial Break-in operations.

Operating parameters

The same limitations apply as for the Attendant Break-In and Network Attendant Service (NAS) Break-In features.

Feature interactions

All of the same feature interactions apply as for the Break-in and Network Attendant Service Break-in features.

The appropriate busy indication is given to a Line Lockout Set which has been broken in on.

Feature packaging

Attendant Break-In Busy Indication and Prevention requires Attendant Break-in/Trunk Offer (BKI) package 127.

Feature implementation

LD 15 – Define break-in Indication and Prevention options.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT	Attendant Console Options.
...		
- OPT	(BIXA) BIXD (BIND) BBIN EBIN	Break-in to external call (allowed) denied. Break-in Indication (denied), Basic Beak-in Indication. Extended Break-in Indication.

Feature operation

For operating procedures, refer to the “Attendant Break-In” on page 237 in this guide.

Attendant Break-In to Inquiry Calls

Contents

This section contains information on the following topics:

Feature description 253
Operating parameters 254
Feature interactions 254
Feature packaging 256
Feature implementation 257
Feature operation 257

Feature description

The Attendant Break-In to Inquiry Calls feature allows an attendant to Break-In to an inquiry call. An inquiry call exists when two stations are established in a simple connection, and one station offers a call transfer to another station. The set making the call transfer becomes the controlling party, and the station receiving the call transfer becomes the active party. The other station is placed on hold and becomes the held party.

The attendant can Break-In to either the controlling or active party, in post-dial or pre-dial operation, by pressing the Break-In (BKI) key. After Break-In has occurred, a Break-In conference is established. All parties receive intrusion tone. While in the Break-In conference, the attendant has consultation status only. The attendant cannot extend a call from the source side.

The attendant cannot Break-In to the held call, to an inquiry call that is in the dialing state or ringing state, or to the active or controlling party if either of them has Warning Tone Denied Class of Service.

To release from the Break-In conference, the attendant presses either the RLS SRC key (to release from source) or RLS DEST key (to release from destination). The inquiry call is restored to its previous state.

Operating parameters

Once in the Break-In conference, the operation of the console Release key is ignored. The operation of the Transfer key (TRN) and Add-on Conference key (AO3/AO6) for proprietary telephones is ignored. For analog (500/2500 type) telephones, a switchhook flash, ground button, or recall operation is ignored.

This feature does not allow the attendant to Break-In to a held party, controlling party while dialing, or the active party during ringing.

The attendant will be unable to Break-In on an inquiry call if either the controlling or active parties has a Warning Tone Denied (WTD) Class of Service.

Feature interactions

Attendant Break-In

All other interactions are the same as for the Attendant Break-In feature.

Attendant Break-In with Secrecy

Attendant Break-In with Secrecy interacts with Attendant Break-In to Inquiry Calls (BIEC) when the desired party has gone on-hook leaving an undesired party off-hook and excluded. BIEC has enhanced the existing BKI feature by giving overflow tone to the undesired party if it is a 500 type set (irrespective of whether the undesired party was involved in an inquiry call or not). BKIS does not change this operation for non-BKIS calls.

BKIS has a choice of options to be given to the undesired party if the desired party goes on-hook while the undesired party is excluded. These are taken from the AOCS options in the Customer Data Block. These options are not given to the undesired party if the undesired party has a call on hold, this only applies to analog (500/2500 type) telephones. The BIEC treatment of giving overflow tone is done instead so that the undesired party can be reconnected to the held party.

Therefore, it is quite possible for analog (500/2500 type) telephones and trunks to get different treatment depending on the circumstances.

The following is a list of treatments for different circumstances:

- Existing BKI BIEC disconnects undesired parties when the desired party goes on-hook, except for analog (500/2500 type) telephones where overflow is given. Therefore proprietary telephones and trunks are disconnected.
- BKIS will give either overflow, transfer to attendant, or disconnect treatment to analog (500/2500 type) telephones or trunks. Proprietary telephones are disconnected.

Automatic Call Distribution Agent/Supervisory Consultation Calls

A consultation Call from an Automatic Call Distribution (ACD) agent to the supervisor, invoked on the Supervisor key on the agent set, is not considered an inquiry call and is not affected by the Break-In to Inquiry Calls feature.

Automatic Hold

A consultation call on a proprietary telephone, using a second DN along with Automatic Hold, is not treated as an inquiry call. The consultation call may be broken-in to, but the call held on the first DN is not involved in the Break-In.

Call Forward All Calls/Call Forward No Answer/Call Forward by Call Type/Do Not Disturb

The operation of these features are overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

Call Forward All Calls/Call Forward No Answer/Make Set Busy/Do Not Disturb

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Call Forward All Calls/Call Forward No Answer/Make Set Busy/Do Not Disturb that may be applied to the set.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Do Not Disturb

The operation of Do Not Disturb is overridden on an analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Do Not Disturb that may be applied to the set.

Held Call Clearing

Held Call Clearing takes precedence over Break-In to Inquiry Calls.

Misoperation During Transfer/Inquiry

Break-In to Inquiry Calls takes precedence over Misoperation During Transfer/Inquiry on a proprietary telephone inadvertently placed on-hook during a Break-In conference, for those cases where the misoperation treatment differs.

Feature packaging

Attendant Break-In to Inquiry Calls requires the Attendant Break-In/Trunk Offer (BKI) package 127.

Feature implementation

LD 12 – Assign Break-In (BKI) to a console key.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	xxxx	Attendant Console type, where xxxx is: ATT, 1250, or 2250.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx BKI	Key number; Break-In.

Feature operation

For operating procedures, refer to the *"Attendant Break-In" on page 237* feature module in this guide.

Attendant Break-In to Lockout Set Denied

Contents

This section contains information on the following topics:

Feature description 259
Operating parameters 259
Feature interactions 259
Feature packaging 260
Feature implementation 260
Feature operation 260

Feature description

The Break-In to Lockout Set Denied (BKLS) enhancement provides an option to prevent an attendant from breaking in on a analog (500/2500 type) telephone that is in a line-lockout state. This feature is applied on a customer basis and has precedence over other line-lockout or Break-In functions.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

BKLS takes precedence over any other feature dealing with Break-In to a line lockout state.

Feature packaging

Attendant Break-In to Lockout Set Denied requires Attendant Break-In/ Trunk Offer (BKI) package 127.

Feature implementation

LD 15 – Allow or deny the Break-In to Line Lockout Set feature:

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT	Attendant Console Option
...		
- OPT	(BLA) BLD	Break-In to Line Lockout Set (allowed) denied.

Feature operation

For operating procedures, refer to “Attendant Break-In” on page 237 in this guide.

Attendant Break-In with Secrecy

Contents

This section contains information on the following topics:

Feature description 261
Operating parameters 262
Feature interactions 262
Feature packaging 264
Feature implementation 264
Feature operation 266

Feature description

The Attendant Break-In with Secrecy (BKIS) feature enhances the capabilities of the Attendant Break-In feature. When a Break-In conference (attendant, desired party, and undesired party) is established and intrusion tone is provided, the attendant can press the Break-In (BKI) key again to exclude the undesired party and talk to the desired party without the intrusion tone.

BKIS applies to both pre-dial and post-dial Break-In operations. In a post-dial situation, the attendant dials the desired party before pressing the BKI key. Whereas in a predial case, the attendant presses the BKI key prior to dialing the digits of the desired party.

BKIS operates in a stand-alone environment and within a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN) environment.

In an MCDN ISDN environment, BKIS is an enhancement of Network Attendant Service (NAS) Break-In (BKI). Please refer to *ISDN Primary Rate Interface: Features* (553-3001-369) for more information regarding Network Attendant Service Break-In.

Operating parameters

The same feature requirements apply as for the Break-In feature.

Within an ISDN environment

- All conditions for NAS Break-In must be met.
- In order for this feature to operate correctly over the network, all nodes connected to the attendant must have Break-In software equipped.

In all cases, when displays are equipped, the information displayed is consistent with current operation (that is, when connected to only one party, the display shows the number and name, if equipped and configured, of that party, and when connected to more than one party, the display is blank).

Feature interactions

Other than the interactions described below, the feature interactions are the same as for the Break-In and NAS Break-In features.

Break-In to Enquiry Calls

Break-In with Secrecy interacts with Break-In to Enquiry Calls (BIEC) when the desired party has gone on-hook leaving an undesired party off-hook and excluded. BIEC has enhanced the existing BKI feature by giving overflow tone to the undesired party if it is a 500 type set (irrespective of whether the undesired party was involved in an enquiry call or not). BKIS does not change this operation for non-BKIS calls.

BKIS has a choice of options to be given to the undesired party if the desired party goes on-hook while the undesired party is excluded. These are taken from the AOCS options in the Customer Data Block. These options are not given to the undesired party if the undesired party has a call on hold. This only applies to analog (500/2500 type) telephones. The BIEC treatment of giving overflow tone is done instead so that the undesired party can be reconnected to the held party.

Therefore, it is possible for analog (500/2500 type) telephones and trunks to get different treatment depending on the circumstances.

The following is a list of treatments for different circumstances:

- Existing BKI BIEC disconnects undesired parties when the desired party goes on-hook, except for analog (500/2500 type) telephones where overflow is given. Therefore proprietary telephones and trunks are disconnected.
- BKIS will give either overflow, transfer to attendant, or disconnect treatment to analog (500/2500 type) telephones or trunks. Proprietary telephones are disconnected.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Multi-Party Operation

For Multi-Party Operation (MPO), the operation of features, for example, going on-hook and releasing from a call during the BKIS conference between the attendant and the desired party, takes precedence over MPO operations for those cases where the treatment differs from that defined by the customer.

All network nodes must have MPO software, with identical Multiple-party Operation (MPO) options. Otherwise, MPO options in the desired party's node have precedence.

Pertaining to MPO options, if the undesired party is not located on the same node as the desired party, the undesired party is considered as an external party on the desired party node.

Music

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Network Attendant Service (NAS)

The BKIS feature operates in a networking environment with regard to the NAS Break-In feature operations and limitations. Please refer to *ISDN Primary Rate Interface: Features* (553-3001-369) for further information on the Network Attendant Service (NAS) feature.

Secrecy Enhancement

The source and destination parties cannot be joined together on the attendants conference bridge if BKIS is active. This is consistent with the existing Break-In feature.

Feature packaging

Attendant Break-In with Secrecy requires the following packages:

- Attendant Break-In (BKI) package 127.
- In an MCDN ISDN environment, ISDN Basic (ISDN) package 145, ISDN Supplementary Features (ISDNS) package 161, and Network Attendant Service (NAS) package 159 are required.
- Multi-Party Operations (MPO) package 141 is optional. If used in an MCDN ISDN environment, all nodes must be equipped with the MPO package.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 12 – Assign Break-In to a key on the Attendant Console.

2 LD 15 – Modify Multi-Party Operations data if MPO package 141 is equipped.

LD 12 – Assign Break-In to a key on the Attendant Console.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	xxxx	Attendant Console type, where: xxxx is: ATT, 1250, or 2250.
...		
AADN	...	
KEY	0-19 BKI	Key number assigned to Break-In.
...		

LD 15 – Modify Multi-Party Operations data if MPO package 141 is equipped.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	MPO	Multi-Party Options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
MPOP	(NO) YES	Multi-Party Operations.
...		
- FMOP	(NO) YES	Flexible Misoperation Options.

- - AOCS	xxxyyy	All Other Cases, where: xxx is for internal calls and yyy or ATN is for external calls.
	AAR AAR	The transferring station is re-rung. If the transferring station fails to answer, the transferred station is routed to the attendant.
	ATN(ATN)	Attendant
	DAR DAR	The transferring station is re-rung. If the transferring station fails to answer, the transferred station is disconnected.
	(DIS) DIS	Disconnect
	OVF OVF	Overflow
	STD STD	Standard
...		

Feature operation

Break-In to two-party connection

The following sections describe a post-dial Break-In. For pre-dial Break-In, Break-In is done on the Source of the attendant and there is no party A calling the attendant, but the BKIS operation is identical.

The scenario is the following:

> Party A calls the attendant. The attendant calls party B who is talking to party C. The attendant presses the BKI key to intrude into the conversation. At this point, the attendant and both parties B and C are in conversation with intrusion tone provided, while party A is on HOLD (with music if EMUS, package 119, is equipped).

Break-In "Allowed"

This situation will arise when party A is an external call and Camp-on or Call Waiting is possible at the wanted station B. At this point, the BKI, Exclude Source (EXCL SRC) and Exclude Destination (EXCL DEST) indicators are active (lamps are lit or Liquid Crystal Display [LCD] is on), and the following actions can occur:

Request the unwanted party to terminate

The attendant may request the unwanted party to terminate. A positive response will terminate the conference that included the attendant and intrusion tone. This is a current BKI operation.

Request the wanted party to terminate

The attendant may request the wanted party to terminate the call. The party disconnects, terminating the BKI conference. This is a current BKI operation.

Attendant presses Release Destination key

The attendant may press the RLS DEST key to release the call. This action terminates the conference and the original call is reestablished as it was prior to Break-In. The Source party A is connected to the Attendant. This is a current BKI operation.

Attendant presses Exclude Destination key

The attendant may press the EXCL DEST key to return to the incoming call. The intruded parties keep receiving the intrusion tone. This is a current BKI operation.

Attendant presses Release key

The attendant may press the Release (RLS) key to apply Camp-on. This is a current BKI operation.

Attendant presses Break-In key again

The BKIS feature allows the attendant to press the BKI key ***again*** in order to exclude the undesired party C (who continues to hear intrusion tone) and to talk directly to the desired party B without intrusion tone. The BKI indicator, which was active, flashes at 60 impulses per minute (ipm).

> ***Note:*** When the attendant presses the BKI key a second time with the Break-In conference excluded, is not activated (that is, if the Break-In conference is on the destination but the attendant is talking on the source, secrecy cannot be activated).

From this point, the following attendant operations can occur:

Attendant actions

Break-In

The attendant presses the flashing BKI key. In this case, party C, which was excluded, is brought back into conversation with the attendant, party B, and intrusion tone. The BKI indicator reverts to an active state. The situation reverts to a normal BKI conference with intrusion tone.

In other words, the lit BKI key can be used to exclude the unwanted party from the BKI conference and the flashing BKI key can be used to reestablish the BKI conference (with intrusion tone).

Exclude Destination

The attendant presses the EXCL DEST key to return to the incoming call. The attendant is connected to the source party. The unwanted party B and the wanted party C are reconnected with intrusion tone. The EXCL SRC indicator is now off and the EXCL DEST lamp and the BKI indicators are active. The operation of the EXCL DEST key has the same effect as for a normal BKI conference situation, as described previously.

Release

The attendant presses the RLS key to apply Camp-on. If Camp-on or Call Waiting is available, parties B and C are reconnected and party A is released and either Camp-on or Call Waiting is applied to the wanted party A. The BKI indicator is off. If Camp-on or Call Waiting is not available, the operation of the RLS key causes secrecy to be turned off and the situation to go back to the Break-In conference with intrusion tone. The loop can only be released by pressing the RLS DEST key, leaving the source connected to the attendant. The operation of the RLS key has the same effect as for a normal BKI conference situation, as described previously.

Release Destination

The attendant presses the RLS DEST key. The BKI, EXCL SRC, and EXCL DEST indicators are off and party A is connected to the attendant. Party B (desired) and party C (excluded party) are reconnected.

Undesired party action

Party C (undesired party) goes on-hook and is disconnected. Then the BKI indicator goes off and the attendant treats the call as a normal two-party connection. The attendant is talking directly to party B (desired party) and can press the RLS key to extend the call.

Desired party action

At this point, if party B (controlling party) goes on-hook, the treatment depends upon the Customer Data Block (LD 15) Multi-party Operations (MPO) Flexible Misoperation Options (FMOP) All Other Cases (AOCS) settings if the undesired party is a trunk or 500-type set and MPO package 141 is equipped. If the MPO package is not equipped, internal calls will be disconnected, while external calls will be rerouted to the attendant.

The following shows what happens to 500-type sets or trunks depending on the AOCS options:

AOCS set to AAR for party C

If AOCS is set to AAR for party C, then party C is routed to the attendant and party B is re-rung by the attendant. BKI indicator goes off and a simple call is set up between attendant and party B when B answers.

AOCS set to ATN for party C

If AOCS is set to ATN for party C, then party C is routed to the attendant while B is re-rung by the attendant. The BKI indicator goes off and the attendant hears ring back and the DEST indicator winks at 30 ipm. The attendant can extend the call as normal.

AOCS set to DAR for party C

If AOCS is set to DAR for party C, then party C is disconnected and party B is re-rung by the attendant. The BKI indicator goes off and when B answers a simple call exists between the attendant and party B.

AOCS set to DIS for party C

If AOCS is set to DIS for party C, then C is disconnected and party B is re-rung by the attendant. The BKI indicator goes off and the attendant hears ringback and the DEST indicator winks at 30 ipm. The attendant can then extend the call as normal.

AOCS set to OVF for party C

If AOCS is set to OVF then overflow tone is given to party C and party B is re-rung by the attendant. The BKI indicator goes off, the attendant hears ringback, and the DEST indicator winks at 30 ipm. The attendant can then extend the call as normal.

AOCS set to STD for party C

If AOCS is set to STD for party C, the treatment is the same as default for the AOCS option. If party C is internal, then DIS option applies to party C, and if party C is external, then ATN option applies to party C.

Break-In 'Consultation Only'

This console state indicates that the attendant has been allowed to Break-In to the desired party's call; however, the attendant will not be able to extend the originating call. This situation will occur under any of the following conditions:

- An internal call is on the source port of the Attendant Console.
- The attendant originated the call. In this case, the source indicator will be used instead of the destination indicator to provide status information (predial situation).
- An external call is on the source and neither Camp-on nor Call Waiting is possible at the wanted station (that is, Camp-on or Call Waiting not possible or the station already has a call camped on).
- The desired station is busy with Call Forward active and the attendant initiated a predial Break-In.

The BKI and the EXCL SRC indicators are active, the DEST indicator is flashing. At this point, the attendant is not allowed to press the RLS key to extend the originating call, party A. The operation of the RLS key is ignored. This is a current BKI operation.

The attendant may press the BKI key to exclude party C and talk directly to party B, as described under the Attendant actions section. The BKI and DEST indicators are flashing. While in this state, the attendant is not allowed to press the RLS key to extend the originating call, party A. The operation of the RLS key causes the secrecy to be turned off and the situation to revert to a Break-In conference. The other operations described in the Attendant actions section are available.

Break-In to a conference

Party A (either internal or external) calls the attendant, the attendant calls party B who is involved in a conference call with parties C and D. The attendant presses the BKI key to intrude into the conversation. At this point, the attendant, party B and all the original conferees are in conversation with intrusion tone provided, while party A is on HOLD. The BKI and EXCL SRC indicators are active. The DEST indicator is flashing and the BKI status is 'Consultation Only'.

At this point, the attendant may press the BKI key to talk directly to party B without intrusion tone. The Break-In indicator flashes at 60 ipm. The original conference is excluded from party B (the other parties in the conference remain connected without intrusion tone). Party A is still excluded on the attendant loop and the attendant is talking directly to party B without intrusion tone.

While in this state, the following situations can occur:

Attendant actions

Break-In

The attendant may press the flashing BKI key. The original conference is reestablished with intrusion tone. The BKI indicator reverts to active.

Exclude Destination

The attendant may press the EXCL DEST key to return to the incoming call. The original conference is reestablished and party A is connected to the attendant.

Release

The attendant is not allowed to extend the original call to the wanted party B by pressing the RLS key. The operation of the RLS key causes the secrecy to be turned off and the situation reverts to a Break-In conference.

Release Destination

The attendant may press the RLS DEST key. The BKI, EXCL SRC and EXCL DEST indicators are off and party A is reconnected to the attendant. The original conference (B, C, and D) is reestablished.

Undesired party action

All but one of the conferees (C or D) go on-hook. The last undesired party will start getting the intrusion tone once again. The situation reverts to the previously described operation (See "Undesired party action" on page 269).

Desired party action

At this point, if party B goes on-hook, party B is re-rung by the attendant and the conferees are left in conference without party B and without intrusion tone. The BKI indicator goes off, the attendant hears ringback tone, and the DEST indicator winks at 30 ipm. The attendant can extend the call as normal.

Table 8 is a summary of possible Break-In situations and indications.

Table 8
Summary of possible Break-In situations and indications (Part 1 of 2)

State	Operation	SRC or DEST Indicator	Break-In Indicator	Tone
1. Allowed	a) post-dial	ACTIVE	ACTIVE	intrusion
	predial	ACTIVE	ACTIVE	busy
	b) post-dial	ACTIVE	OFF	none
	predial	ACTIVE	ACTIVE->OFF	override
2. Consultation Only	a) post-dial	FLASH	ACTIVE	intrusion

Table 8
Summary of possible Break-In situations and indications (Part 2 of 2)

State	Operation	SRC or DEST Indicator	Break-In Indicator	Tone
	b) predial	FLASH	ACTIVE	busy
3. Temporarily Denied 1		FLASH	FLASH	busy override if override is involved
4. Temporarily Denied 2	a) post-dial only	FLASH	WINK	overflow
	b) predial	FLASH	WINK	busy or ring back
	(then post-dial)	FLASH	WINK	intrusion
5. Denied		FLASH	OFF	overflow
6. Break-In	a) post-dial	WINK	OFF	ringback
Ignored station is rung	b) Predial	WINK	OFF	ringback
7. Invalid	post-dial or predial	OFF	OFF	overflow
8. Break-In with Secrecy	after post-dial or predial, active BKI key is pressed	ACTIVE or FLASH	FLASH	no tone

Table 9 is a summary of possible Break-In situations and actions.

Table 9
Summary of possible Break-In situations and actions

Condition of called DN	Action
1. Established call, Call Waiting or Camp-on allowed, Multiple Appearance DN. Lockout (if not denied).	Break-In allowed, connection established. Connection is made.
2. Attendant dialing on SRC, internal call on SRC, CWT or Camp-on not available, desired party in conference, Call Forward active on set.	Connection is made for the attendant only.
3. Tones, ringing, dialing, blocking, Override, Camp-on, Hold, talking to another attendant, Call Transfer, WTD on undesired party.	Release DEST, wait and repeat.
4. Make Set Busy, Do not disturb.	Predialing operation possible.
5. Warning tone denied on desired party, maintenance busy.	Break-In impossible.
6. Station is idle.	Station is rung, station not affected.
7. Invalid numbers.	Break-In impossible.
8. The previous status was “Allowed” or “Consultation Only”. SRC or DEST indicator was active ('Allowed') or flashing ('Consultation Only').	Undesired party is excluded and the attendant is talking to the wanted party.

Attendant Busy Verify

Contents

This section contains information on the following topics:

Feature description 275
Operating parameters 276
Feature interactions 276
Feature packaging 278
Feature implementation 278
Feature operation 280

Feature description

Attendant Busy Verify allows the attendant to establish a connection with any apparently busy DN to verify that the DN is actually busy and in working order. This feature can also be used to connect with a busy station if an emergency situation requires call interruption by the attendant.

When Busy Verify is active, a 256 millisecond burst of interrupted tone is sent every six seconds to indicate the presence of the attendant. The attendant can Busy Verify only those stations with Warning Tone Allowed Class of Service.

When a station is involved in a conference, the attendant can verify whether the station is busy even if it has Warning Tone Denied Class of Service.

An attendant can also use either the Release Source or Release Destination key on the console to release one of the parties involved in a Busy Verify conference.

Operating parameters

The system must be equipped with a conference loop.

If equipped, the Busy Verify key must be assigned to key 0 of the console flexible feature strip.

Feature interactions

Attendant Break-In

The attendant can use the Break-In key instead of Busy Verify to break in to an established call. Attendant Break-In simplifies this process.

Automatic Redial

Attendant Break-In and Attendant Busy Verify are not permitted on proprietary sets that are used for an Automatic Redial (ARDL) call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Call Forward All Calls

If the DN is call forwarded to the Attendant Console, the attendant will receive a click followed by silence.

Call Forward Busy Hunting

Call Forward Busy and Hunting do not affect Busy Verify.

Call Forward/Hunt Override Via Flexible Feature Code

Using Call Forward/Hunt Override Via FFC after activation of Barge-in, Busy Verify or Break-in is not allowed. Attempts will be canceled and overflow tone will be returned.

Using post-dial Break-in after dialing the Call Forward/Hunt Override FFC is possible after encountering a busy set, if Break-in is enabled.

Call Forward, Internal Calls

When the attendant is using this feature to call a telephone that is Internal CFW active, the call will not receive Internal CFW treatment.

Charge Account and Calling Party Number

A charge account number cannot be entered when Attendant Barge-In or Attendant Busy Verify is active. Barge-In cannot be used to connect to a trunk after an account number has been entered.

China – Attendant Monitor

When China (CHINA) package 285 is equipped, the normal operation of Busy Verify changes. The repeatable tone is now configurable with the (TOA)/TOD option.

If an attendant is monitoring a DN, a second attendant defined for the same customer will be blocked from Busy Verifying any party involved in the monitored call.

If an attendant is Busy Verifying a DN, a second attendant defined for the same customer will be blocked from monitoring any party involved in the monitored call.

Conference

Conference Control cannot be activated if an attendant has used Busy Verify during a conference that involves a trunk.

Direct Inward System Access

Attendant Busy Verify applies only to DNs within the system. If an attendant tries to use the feature to enter a Direct Inward System Access DN, overflow tone is returned.

Group Hunt

An attendant is not allowed to busy-verify when dialing a Pilot DN directly.

Intercept Computer Dial from Directory - Pre-dial Operations

It is possible for an attendant to override call forward on a set in the following manner:

- Press an idle loop key, and press the Break-in key on the Attendant Console.
- Dial an extension DN from the Intercept Computer.

Music, Enhanced

When the attendant attempts to Busy Verify a telephone receiving Music, the Music is removed. When the attendant releases, Music is returned.

On Hold on Loudspeaker

It will not be possible to Busy Verify into a call on loudspeaker as it is effectively on hold at the set.

Periodic Camp-on Tone

The Periodic Camp-On Tone has precedence over Busy Verify intrusion tone.

Uninterrupted Line Connections

Attendant Busy Verify cannot be applied to stations with a Warning Tone Denied Class of Service.

Feature packaging

Attendant Busy Verify is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 12 – Add/change a Busy Verify key on Attendant Consoles.

2 LD 10 – Allow/deny Warning Tone Class of Service for analog (500/2500 type) telephones.

3 LD 11 – Allow/deny Warning Tone Class of Service for proprietary telephones.

4 LD 14 – Allow/deny Warning Tone Class of Service for trunks.

LD 12 – Add/change a Busy Verify key on Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	0 BVR	Add a Busy Verify key.

LD 10 – Allow/deny Warning Tone Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Allow or deny warning tone.

LD 11 – Allow/deny Warning Tone Class of Service for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Allow or deny warning tone.

LD 14 – Allow/deny Warning Tone Class of Service for trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	COT DID FEX RAN TIE WAT	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Allow or deny warning tone.

Feature operation

To verify a busy DN, follow these steps:

1. Select an idle loop key.
2. Press **Busy Verify**.
3. Dial the DN of the station.

   If the DN is idle, press **Signal Source** to ring the station.

   Possible results are the following:

   — silence (DN is idle and working)

   — conversation (DN is busy and working)

   — fast busy (station is disabled or has Warning Tone Denied Class of Service).

4. Press the **Rls** key to disconnect from the call.

An enhancement to the Busy Verify feature offers the following functionality. Party A is on a call with Party B. The attendant

1 Selects an idle loop key.

2 Presses **Busy Verify**.

3 Dials Party A and creates a Busy Verify conference between Party A, Party B, and the attendant.

The use of the **Rls DEST** and **Rls SOURCE** keys are allowed at this point as follows:

- The attendant can press the **Rls DEST** key to release Party A from the Busy Verify conference or
- The attendant can press the **Rls SOURCE** key to release Party B from the Busy Verify conference.

Attendant Call Selection

Contents

This section contains information on the following topics:

Feature description 283
Operating parameters 284
Feature interactions 284
Feature packaging 284
Feature implementation 284
Feature operation 284

Feature description

All calls to the attendant, with the exception of slow-answer recalls, are automatically queued in order of arrival. The attendant can answer a call in two ways:

- Calls can be answered in the order received, regardless of call type, using the Loop key (LPK).
- A particular call type can be answered before other calls in the queue by manually selecting the appropriate Incoming Call Indicator (ICI) key.

The first call presented to an idle console is indicated by the appropriate ICI lamp. All subsequent calls are indicated by the Calls Waiting lamp only until the first call is released. All appropriate ICI lamps will then light, and an attendant may select a specific incoming call type by pressing the appropriate ICI key.

If a customer has multiple consoles, the first call in queue is presented to the first idle console.

Operating parameters

The maximum number of ICI lamps per Attendant Console is 20. All consoles associated with a customer have the same ICI assignments.

Feature interactions

Attendant Incoming Call Indicators

The ICI feature is used with the Attendant Call Selection feature to recognize, answer, and process incoming calls.

Feature packaging

This feature is included in base System Software.

Feature implementation

No change to existing configuration is required for the Attendant Call Selection feature.

Note: To implement ICI, see "Attendant Incoming Call Indicators" on page 323 of this document.

Feature operation

The attendant can answer a call by:

- pressing the Loop key to answer calls in the order received, or
- pressing the appropriate ICI key to answer a call by call type.

Attendant Calls Waiting Indication

Contents

This section contains information on the following topics:

Feature description 285
Operating parameters 286
Feature interactions 286
Feature packaging 286
Feature implementation 286
Feature operation 288

Feature description

Call Waiting on the console gives the attendant an indication of the number of calls in the console queue and the length of time they have been waiting to be answered. Each console is equipped with a Call Waiting indicator. The indicator is dark when no calls are waiting in the queue. The indicator is steadily lit when one or more calls are waiting. The indicator flashes when the number of waiting calls exceeds the customer defined threshold, or when a call has been waiting longer than the specified number of seconds.

The two thresholds that control the lamp states are defined in the Customer Data Block. The time delay threshold can be specified from 0 to 511 seconds in multiples of two seconds. The number of calls threshold can be specified from 0 to 255. If zero is specified, this aspect of the Call Waiting feature is not operational.

An option is also provided to supply a two-second buzz to notify the attendant when the first call enters the queue or when the Call Waiting lamp changes from steadily lit to flashing, or both.

If the threshold has been exceeded and the Call Waiting indicator is flashing, it changes to steadily lit when the threshold is no longer exceeded by either number of calls or time delay.

Operating parameters

If neither the time delay or number of calls thresholds are defined, the Call Waiting lamp state will not change from steadily lit to flashing.

Feature interactions

Call Park on Unsupervised Trunks

If all the attendants are busy and a Call Park Recall occurs, the recall is placed in the calls waiting queue. If the recalled station is busy when the recall occurs, the Disconnect Timer (DCTI) temporarily suspends timing until the recall is presented. After the recall is presented, the Disconnect Timer continues timing for the remainder of the period.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Define Call Waiting thresholds and indications for a customer.

2 LD 12 – Add/change a Display Calls Waiting key on an Attendant Console.

LD 15 – Define Call Waiting thresholds and indications for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CWUP	(NO) YES	Call Waiting Queue Update. (Do not) automatically notify Attendant Console (M2250) when the number of calls waiting in queue changes.
- CWCL	(0)-255 (0)-255	Call Waiting Call Limit. Lower and upper bound of the threshold for the number of calls waiting (the default is 0).
- CWTM	(0)-511 (0)-511	Call Waiting Time. Lower and upper bound of the threshold for the time calls are waiting (the default is 0).
- CWBZ	 (NO) YES (NO) YES	Call Waiting Buzz (Disable) enable a buzz to the attendant when either the CWCL or CWTM thresholds are exceeded. (Disable) enable a buzz to the attendant when the first call enters the queue.

LD 12 – Add/change a Display Calls Waiting key on an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx DCW	Add a Display Calls Waiting key. xx = 0-9 for QCW or M1250 Attendant Console. xx = 00-19 for M2250 Attendant Console.

Feature operation

If CWUP (notify change in Calls Waiting status) is set to YES in LD 15, the number of calls waiting are displayed on the M2250 console. If CWUP is set to NO, the attendant must press the Display Calls Waiting (DCW) key to display the number of waiting calls.

Attendant Clearing during Night Service

Contents

This section contains information on the following topics:

Feature description 289
Operating parameters 290
Feature interactions 290
Feature packaging 291
Feature implementation 291
Feature operation 292

Feature description

When an Attendant Console is placed in Night Service, the Attendant Clearing during Night Service feature causes all active calls or calls being held on Loop keys to be cleared and given a customer-defined treatment. One of the following treatments can be selected:

- internal calls are disconnected, and external calls are routed to the Night Directory Number (DN)
- all calls are routed to the Night DN
- no clearing

An external call is defined as a call involving at least one external party. The definition of a external party is the same as used for the Multi-Party Operations (MPO) feature. Any CO, DID, or TIE trunk (incoming or outgoing) connected to the system is considered an external party, regardless of the way the connection is established.

Operating parameters

Attendant Clearing during Night Service is offered as part of the Multi-Party Operations feature.

Feature interactions

AC15 Recall: Timed Reminder Recall

If Attendant Clearing During Night Service is active and there is a call being extended over an AC15 TIE trunk, when the attendant goes into Night Service, the transfer is completed and the feature is activated.

If there is an AC15 recall presented to the attendant and it goes in Night Service, the recall is put in the attendant queue.

If an AC15 recall has been answered by the attendant and it goes in Night Service, the call is removed from the attendant port and the feature is activated again.

Night Service Enhancements

The Night Service Enhancements features take precedence over Attendant Clearing during Night Service.

Scheduled Access Restriction

Attendant Clearing during Night Service should be equipped with Scheduled Access Restriction (SAR). When Night Service is in effect, the only operations that can be performed from Attendant Consoles, which are members of a SAR group, are:

- release any existing calls, or
- dial one of the following SAR Flexible Feature Codes:
  - Scheduled Access Disable (SADS)
  - Scheduled Access Enable (SAEN)
  - Scheduled Access Lock (SALK), or
  - Scheduled Access Unlock (SAUN).

Feature packaging

The Attendant Clearing during Night Service feature is packaged as part of the Multi-Party Operations (MPO) package 141.

Feature implementation

Task summary list

The following task is required:

LD 15 – Configure the Attendant Cleaning during Night Service feature at the ACNS prompt.

LD 15 – Configure the Attendant Cleaning during Night Service feature at the ACNS prompt.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Modify existing data.
TYPE:	MPO	Multi-Party Options
...		

MPOP	YES	Multi-Party Operations options.
...		
- FMOP	YES	Flexible Misoperation Parameters.
- - RCY2	...	
- - ACNS		Attendant Clearing during Night Service.
	(NO)	Attendant will not be cleared (the default).
	ALL	All calls will be routed to the Night DN.
	EXT	External calls will be routed to the Night DN, while internal calls will be disconnected.
	<CR>	Previously defined value not changed, or set to default if response to REQ was NEW.

Note: This overlay is modified to output the Attendant Clearing during Night Service (ACNS) prompt as part of the MPO group of prompts. The ACNS prompt will only appear if the MPO package is equipped and the response to both MPOP and FMOP is YES. The ACNS prompt will accept a response of either NO, ALL, EXT or a carriage return (<CR>).

Feature operation

A customer is put into Night Service manually, by pressing the NITE key on the Attendant Console or having all Attendant Consoles activate Position Busy, or automatically, by the Scheduled Access Restrictions (SAR) or Attendant Forward No Answer (AFNA) features. When Night Service is activated, all calls or selected calls associated with the attendant will be given treatment according to the feature option defined in the Customer Data Block (LD 15) as part of the Multi-Party Operations (MPO) options.

The sections following describe the treatments given to different call types.

Established Calls

Single Party Call

Both the incoming or outgoing single party call, (not associated with another call on the attendant) established on the attendant Source (SRC) or Destination (DEST) sides will be routed to the Night DN.

Two Party Call – Ready to Extend

When a call is being extended, a call excluded on the SRC side and an outgoing call established on the DEST side, the call will be extended provided it is allowed as if the Release (RLS) key is pressed. If it is not allowed due to access restriction, the outgoing call on the DEST side will be disconnected and the call on the SRC side will be routed to the Night DN.

Conference Call on Source

If a conference call is established on the attendant SRC, the attendant will be excluded from the conference and disconnected as if the Release Source (RLS SRC) key were pressed.

Conference Call on Destination

If a conference call is established on the attendant DEST (Break-In conference) the attendant will be excluded from the conference and disconnected as if the Release Destination (RLS DEST) key were pressed.

Non-established Calls

Any call in the dialing state on either the SRC or DEST side will be disconnected.

Any call in the ringing state or receiving any tone on either the SRC or DEST side will be dropped or disconnected as if the RLS SRC or DEST key was pressed.

If the call in the ringing, dialing or receiving tone state is on the DEST side, and there is an established call in the EXCLUDE state on the SRC side, the SRC party will be rerouted to the Night DN.

Calls held on the console Loop keys

Any established calls being held on a Loop key will be released and calls extended where possible as described in the Established Calls section, or routed to the Night DN.

When a held call is routed to the Night DN, the held party, which is listening to silence or Music on Hold if available, will receive Ringback Tone. If the Night DN is not idle, the call will be placed in the Call Waiting queue.

Attendant Consoles

Contents

This section contains information on the following topics:

Feature description 295
Operating parameters 302
Feature interactions 302
Feature packaging 302
Feature implementation 302
Feature operation 307

Feature description

Attendant Consoles assist in placing and extending calls into and out of the system. The operator of an Attendant Console is known as the attendant. The consoles provide the attendant with many unique features that increase the speed and ease of call processing.

This feature module provides an overview of the Attendant Consoles and a description of the basic software capabilities and associated service changes. Additional information regarding attendant-related software features can be found in other feature modules in this document.

Table 10 describes the Attendant Consoles that are available with the system.

Table 10
Attendant Console types

Console Type	Description
M1250	Console with a 4-line, 40-character wide alphanumeric Liquid Crystal Display
M2250	Digital console with a 4-line, 40-character wide alphanumeric Liquid Crystal Display

Both the M1250 and M2250 consoles have a four line LCD alphanumeric display, each line 40 characters wide, which displays the information presented in Table 11.

Table 11
LCD alphanumeric display information

Line	Display information
1	Displays the time and date.
2	Displays call source information.
3	Displays call destination information.
4	Displays console status information.

Directly below the display screen is a horizontal row of keys that provide the Position Busy, Night Service, Signal Source, and Signal Destination functions.

The M1250 and M2250 consoles have five vertical keystrips that provide the following functions. The Attendant Consoles have a digit display at the top of the console and a dial pad below the display. Five vertical keystrips on the console provide access to the functions described in this section.

Vertical keystrip 1

This keystrip at the far left on the console is utilized for Trunk Group Busy (TGB) keys. The attendant can deny stations access to a trunk route by pressing the associated Trunk Group Busy key. Additionally, the lamps associated with Trunk Group Busy keys provide the visual indication of the status of the trunks within the route (See Table 12).

Table 12
Visual Indication of the status of the trunks within the route

Visual Indication	Status of the trunks within the route
Dark	Some of the trunks in the route are idle.
Flashing	All of the trunks in the route are busy.
Steadily lit	The attendant has taken control of the route.

The basic Attendant Console has 10 Trunk Group Busy keys. If an add-on module is installed, there are 16 Trunk Group Busy keys.

Vertical keystrip 2

This keystrip is used for Incoming Call Indicator keys. The Incoming Call Indicators (ICIs) identify the type of calls in the queue and the status of each particular call type. Three lamp states are associated with each Incoming Call Indicator key (See Table 13).

Table 13
Key lamp states associated with each Incoming Call Indicator key

Lamp state	Status of call type
Dark	No calls of this type are waiting.
Flashing	One call of this type is waiting in queue.
Steadily lit	Two or more calls of this type are queued, or one call has been waiting longer than 20 seconds.

To select a specific type of incoming call, the Incoming Call Indicator key associated with a steadily lit or flashing LED is pressed. The call is removed from the queue and presented to an idle loop key on the Attendant Console.

The basic Attendant Console has 10 Incoming Call Indicator keys. If an add-on module is equipped, the console may have 20 Incoming Call Indicator keys. An Incoming Call Indicator key may be assigned to one or more of the call types listed in Table 14.

Table 14
Incoming Call Indicator key assignments

Key	Mnemonic	Meaning
00-19	CAx	Station Category Number (x = 1-7)
00-19	CFB	Call Forward Busy
00-19	CFN	Call Forward No Answer
00-19	DF0	Dial 0 fully restricted
00-19	DL0	Dial 0
00-19	IAT	Inter-attendant call
00-19	INT	Intercept
00-19	LCT	Lockout
00-19	LD0	Listed DN 0
00-19	LD1	Listed DN 1
00-19	LD2	Listed DN 2
00-19	LD3	Listed DN 3
00-19	MWC	Attendant Message Center
00-19	RLL	Recall
00-19	Rxxx	Route number

Vertical keystrip 3

This keystrip includes the following operating keys:

Release – Allows the attendant to release a call from the console. When the release lamp is lit, it indicates that no incoming calls are being presented to the console.

Loop key/lamps – Allows the attendant to answer and originate calls from the console. The first call in the attendant queue is automatically presented to an idle loop key. Subsequent calls are queued and presented to a loop key when the console becomes idle.

Three lamp indicators, positioned on the upper right-hand side of the keystrip, provide the following information:

- **Two Alarm indicators:** When steadily lit, the minor alarm lamp indicates the system has detected a malfunction that does not affect normal call processing. When the major alarm lamp is steadily lit, the system has detected a malfunction that does not permit normal call processing.
- **Call Waiting indicator:** The Call Waiting lamp indicates the number of calls in the attendant queue and the length of time they have been waiting to be answered. The lamp changes from steadily lit to flashing when waiting calls exceed a certain number, or when a call has been waiting longer than a specified time. The number of waiting calls are displayed by pressing the Display Calls Waiting key, if assigned.

Vertical keystrip 4

This keystrip provides the following fixed feature keys:

Hold – Allows the attendant to hold a call at the console.

Conference – Permits the attendant to set up a conference of up to five conferees, plus the attendant.

Release Destination – Allows the attendant to release the called party from a call held at the console, while holding the calling party.

Release Source – Allows the attendant to release the calling party from a call held at the console, while holding the called party.

Signal Source and Destination – Allows the attendant to recall either party to a call held on the console.

Exclude Destination – Excludes the called party from an established call held at the console, allowing the attendant to speak privately with the calling party.

Exclude Source – Excludes the calling party from an established call held at the console, allowing the attendant to speak privately with the called party.

Volume Control – Allows the attendant to change the volume of alerting signals. Each depression of the key changes the volume of the signal by one step in an eight step range.

Vertical keystrip 5

The optional features listed in Table 15 can be defined on this keystrip.

Table 15
Attendant Console optional feature key assignments (Part 1 of 2)

Key	Mnemonic	Meaning
00	BVR	Busy Verify
01	BIN	Barge-In
00-09	ADL	Autodial
02-09	AWU	Automatic Wake Up
00-09	CHG	Charge Account
00-09	CPN	Calling Party Number
00-09	DCW	Display Calls Waiting
00-09	DDL	Do-Not-Disturb, Individual
00-09	DDT	Display Date
00-09	DPD	Display Destination
00-09	DPS	Display Source
00-09	DTM	Display Time
02-09	EES	End-to-End Signaling
00-09	GND 0-99	Group Do-Not-Disturb

Table 15
Attendant Console optional feature key assignments (Part 2 of 2)

Key	Mnemonic	Meaning
00-09	MCK	Message cancellation
00-09	MDT	Display/Change Date
00-09	MIK	Message indication
00-09	MTM	Display/Change Time
00-09	PAG xxx…x	Paging (xxx…x = route access code)
00-09	PRG	Attendant Administration
00-09	PRK	Call Park
00-09	RDL	Stored Number Redial
00-09	RTC	Routing Control
00-09	SCC xxxx	Speed Call Controller (xxxx = list number)
00-09	SSC xxxx	System Speed Call Controller (xxxx = list number)
00-09	TRC	Malicious Call Trace

The consoles have a Shift key on the fixed feature key strip that provides access to an Options menu. This menu allows the setting of the display screen contrast, buzz tone, language, time and date format, and calls waiting options. Additional information on the Options menu can be found in the *Telephones and Consoles: Description* (553-3001-367).

The Shift key also allows M1250 consoles to have 20 Incoming Call Indicator keys in the regular mode and 16 Trunk Group Busy keys in the shift mode. The M2250 console can have 20 Incoming Call Indicator keys in the regular mode, and 20 Trunk Group Busy keys and an additional ten flexible feature keys in the shift mode. Add-on modules are not required on the M1250 and M2250 consoles to provide the additional key functions.

Attendant Call Party Name Display (CPND) and the Enhanced Busy Lamp Field/Console Graphics Module capabilities may be equipped with the M1250 and M2250 consoles. Please refer to the feature modules in this document for a complete description of these capabilities.

For additional information on Attendant Consoles and associated hardware, refer to the following Nortel Networks Technical Publications (NTPs):

- *Telephones and Consoles: Description* (553-3001-367)
- *Large System: Maintenance* (553-3021-500)

Operating parameters

Refer to the preceding Nortel Networks technical publications.

Feature interactions

Refer to the preceding Nortel Networks technical publications.

Feature packaging

Attendant Console capabilities are included in base system software.

Calling Party Name Display (CPND) package 95 includes Attendant CPND and requires Digit Display (DDSP) package 19.

M2250 Attendant Console (DCON) package 140 requires M2000 Digital Sets (DSET) package 88.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Attendant Console-related prompts and responses.

2 LD 12 – Add an Attendant Console.

LD 15 – Attendant Console-related prompts and responses.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	LDN	Department Listed Directory Numbers.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- LDN0	xxx…x	Listed Directory Number 0.
- LDA0	xx xx… ALL	Attendant Consoles associated with LDN0 (see Note).
- LDN1	xxx…x	Listed Directory Number 1.
- LDA1	xx xx… ALL	Attendant Consoles associated with LDN1 (see Note).
- LDN2	xxx…x	Listed Directory Number 2.
- LDA2	xx xx… ALL	Attendant Consoles associated with LDN2 (see Note).
- LDN3	xxx…x	Listed Directory Number 3.
- LDA3	xx xx… ALL	Attendant Consoles associated with LDN3 (see Note).
TYPE	NIT	Gate opener.
- NIT1	xxx…x	First Night Service DN.
- TIM1	hh mm	Hour and minute of first Night Service DN.
- NIT2	xxx…x	Second Night Service DN.
- TIM2	hh mm	Hour and minute for second Night Service DN.
- NIT3	xxx…x	Third Night Service DN.
- TIM3	hh mm	Hour and minute for third Night Service DN.
- NIT4	xxx…x	Fourth Night Service DN.
- TIM4	hh mm	Hour and minute for fourth Night Service DN.
TYPE	ATT	Attendant Console options.

ATDN	(0) xxx…x	Attendant DN.
- NCOS	(0)-99	Attendant Network Class of Service for all consoles.
TYPE:	CAS	Centralized Attendant Service options.
- CAS	(NO) YES	Change Centralized Attendant Service options.
TYPE:	ANI	Automatic Number Identification.
OPT	(IC1) IC2 (XTG) ITG (LOD) LOA (XDP) IDP (XLF) ILF (SYD) SYA	10 or 20 Incoming Call Indicators. Trunk Group Busy keys not equipped/equipped. (Deny) allow Lockout. Digit Display not equipped/equipped. Lamp Field Array not equipped/equipped. (Deny) allow Secrecy.
- ANAT	xxx x	Attendant Billing number.
- ANLD	xxx…x	ANI listed DN.
TYPE:	ATT	Attendant Console options.
- LFTN	 l s c u c u	Terminal Number. For Large Systems For Small Systems and Succession 1000 systems
- LFTN	 l s c u c u	Terminal Number. For Large Systems For Small Systems and Succession 1000 systems
- LFFD	xxx…x	First DN of Lamp Field Array.
AATT	xxxx	AIOD attendant identifier.
TYPE:	TIM	Timers.
- RTIM	xxxx yyyy zzzz	Recall timers. xxxx = slow answer (0-378). yyyy = Camp-On (0-510). zzzz = Call Waiting (0-510).
- ATIM	(0)-126	Attendant Alternative Answering timer.

ICI	xx yyy	Incoming Call Indicator key assignment. xx = key number. yyy = mnemonic (see Table 14). ***Note:*** Multiple responses can be entered for the same key. To remove an entry, enter xx NUL, then reenter the desired responses. To add an entry, enter the desired response. It will be added to any already existing response.
- AQTT	0-(30)-255	Attendant queue timing threshold in seconds.
TYPE:	ATT	Attendant Console options.
- AODN	xxxx…x	Attendant overflow DN.
TYPE:	PWD	Gate opener.
- ATAC	xxxx	Attendant Administration access code.
TYPE:	ATT	Attendant Console options.
- CWUP	(NO), YES	Call Waiting queue update.
- CWCL	(0)-255 (0)-255	Call Waiting lower and upper thresholds for number of calls in queue.
- CWTM	(0)-511 (0)-511	Call Waiting lower and upper thresholds for time in queue.
- CWBZ	(NO) YES (NO) YES	Buzz when Call Waiting thresholds are exceeded. Buzz when first call enters queue.
- MATT	(NO) YES	Attendant Consoles used as Message Center.
- SPVC	0-63	Attendant number for supervisor console.
TYPE:	AWU	Automatic Wake Up options.
- AWU	(NO) YES X	Enable Automatic Wake Up (X erases AWU information).
- ATRC	(NO) YES	Attendant Recall after failed AWU attempts.

Note: Enter one or more attendant numbers (1-63). Enter ALL to enable this listed DN on all attendants. Precede the attendant number with X to remove.

LD 12 – Add an Attendant Console.

Prompt	Response	Description
REQ	NEW	Add a console.
TYPE	ATT 1250 2250 PWR	Attendant Console. M1250 console. M2250 console. Power TN.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	(DD) SD	Card density.
SETN	l s c u c u	Second TN (must be on same loop as primary TN of Attendant Console).
ANUM	1-63	Attendant number (1-63).
DLEN	(8) 16	Digit display length (the default is 8). Not prompted if TYPE = 1250 or 2250.
SSU	0-4095	System Speed Call user list number.
ICDR	(ICDD) ICDA	(Deny) allow internal call detail.
CPND	(CNDD) CNDA	(Deny) allow Call Party Name Display. Prompted if TYPE is 1250 or 2250.
DNDI	(DNDD) DNDA	(Deny) allow dialed name display.
EBLF	(BLFD) BLFA	(Deny) allow enhanced busy lamp field. Prompted if TYPE is 1250 or 2250.
AADN	xxx…x	Attendant Alternative Answering DN.
KEY	xx aaa	Key number and mnemonic for feature assignments (see Table 15).

Feature operation

Refer to the appropriate Attendant Console User guide for specific operation procedures.

Attendant Delay

Contents

This section contains information on the following topics:

Feature description 309
Operating parameters 310
Feature interactions 310
Feature packaging 310
Feature implementation 310
Feature operation 311

Feature description

The Attendant Delay feature prevents an attendant from performing the following operations during a customer defined period (0 to 14 seconds inclusive) after a call is presented or recalled to the attendant:

- placing the call on hold
- releasing the call
- parking the call
- extending the call
- performing call splitting
- activating paging
- placing a call, if Secrecy or Enhanced Secrecy applies to the presented call or recall

Operating parameters

If Night Service, Attendant Overflow Position, Position Busy, or Attendant Alternate Answering are active, calls presented or recalled to the attendant are automatically routed to a pre-selected station, and are not subject to Attendant Delay.

Feature interactions

Attendant Console Misoperation

Attendant Delay takes precedence over Attendant Console Misoperation.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Enable Attendant Delay.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	TIM	Timers
...		
- ADHT		Attendant Delay on Hold Timer
	(0)-14	Respond to the ADHT prompt with 0 (the default) to leave this feature disabled, or with a value from 1 to 14 seconds for the Attendant Delay timer to enable the feature. This must be done for each customer to be equipped with the feature.

Feature operation

No specific operating procedures are required to use this feature.

Attendant Display of Speed Call or Autodial

Contents

This section contains information on the following topics:

Feature description 313
Operating parameters 313
Feature interactions 314
Feature packaging 314
Feature implementation 314
Feature operation 314

Feature description

With the Attendant Display of Speed Call or Autodial feature, when an attendant uses the Speed Call or Autodial feature to dial a number automatically, the dialed digits are shown on the console display. The speed-call code and the dialed speed-call number are displayed for a speed-call operation. The dialed autodial number is displayed for autodial operation.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

No specific implementation procedures are required to use this feature.

Feature operation

If an attendant presses the **Speed Call** key, the speed-call code and dialed speed call number are shown on the Attendant Console display.

If an attendant presses the **Autodial** key, the dialed autodial number is shown on the Attendant Console display.

Attendant Forward No Answer

Contents

This section contains information on the following topics:

Feature description 315
Operating parameters 316
Feature interactions 317
Feature packaging 319
Feature implementation 320
Feature operation 321

Feature description

The Attendant Forward No Answer (AFNA) feature is comprised of two capabilities. The first allows Direct Inward Dial (DID), Direct Outward Dial (DOD), or Central Office (CO) calls, presented to the attendant and not answered within a customer-defined period of time to be forwarded to another attendant, or, if the customer is in Night Service, to the night DN.

The second capability allows Direct Inward Dial or Central Office calls, presented to a station that is in Night Service, to be disconnected if not answered within the pre-defined ring cycle, or time period. This second capability is called Night Forward No Answer (NFNA).

Two timers are available: the Attendant Forward No Answer timer (AFNT) and the Attendant Forward No Answer Buzz timer (AFBT), both of which are programmed in LD 15.

If the AFBT timer is programmed, when a call is presented to the attendant, the attendant receives a buzz at maximum volume for the duration of the AFBT timer. If the value set for the AFNT timer is higher than that of the AFBT timer, the attendant receives a buzz at normal volume for the duration between when the two timers expire. The AFNT timer can be set between two and 126 seconds. The AFBT timer cannot be set higher than the AFNT timer.

If the attendant does not accept the call before the AFNT timer expires, the attendant is put in Position Busy and the call is relinked to the top of the queue. If all attendants are put in Position Busy, the call can be forwarded using Attendant Overflow Position (AOP) or Night Service if equipped.

When a call is forwarded from the attendant queue to a busy Attendant Overflow Position, the call remains in the queue. If the AOP is idle, the Attendant Forward No Answer timer is started. If the call is not answered before time-out, the AOP is idled. The call is relinked to the top of the queue. If all attendants are in Position Busy, Night Service is activated and the call is transferred to the night DN.

If the night DN is busy, the call is added to the queue, provided the call involves a CO, FEX, WATS, CAS, or CAMA trunk, or was handled by Enhanced Night Service. Other calls, such as TIE or internal calls, are given busy tone.

During Night Service, when a DID or CO trunk call is presented to an idle DN, the Night Forward No Answer (NFNA) ring counter is started. If the call is not answered during the NFNA time cycle, the call is disconnected. Non-DID and non-CO calls ring until the call is answered or the calling party hangs up.

Operating parameters

Attendant Forward No Answer operates in a standalone or networking environment. For networking applications, the transferring and terminating stations can be located on different nodes.

Attendant Forward No Answer does not apply to inter-attendant calls.

Night Forward No Answer (NFNA) and Night Forward No Answer in seconds (NFNS) do not apply to calls waiting in the ACD queue or the Primary Line Directory Number (PLDN) queue.

When Night Forward No Answer times out on an unanswered trunk, the trunk is locked out until the far-end goes on-hook.

The maximum number of ring cycles for Attendant Forward No Answer on an Attendant Overflow Position is 63.

AFNA timing ceases and the volume of the attendant buzzer is set to the original value in the following cases:

- If the attendant answers a call
- If the attendant answers an Automatic Wake-up recall on the AWU key
- If an attendant-extended call is answered on a set during a slow answer recall to the attendant
- If a call waiting call is answered at a set while the attendant is ringing

If a set or trunk disconnects while the attendant is being rung, and the AFNA timing cannot continue on the source or destination side, the volume of the attendant buzzer is set to its original value.

The NFNS timing starts when a DID/DOD/CO call is recalled to the night station, as part of the Recall to Night Station treatment, requeued to the night station as part of the Requeueing of Attendant Presented Calls treatment, or rerouted to the night station as part of the Attendant Clearing During Night Service treatment.

If both the Disconnect Timer (DCTI) of the Periodic Clearing feature and NFNA or NFNS are defined, the first one which expires will disconnect a DID or CO call.

Feature interactions

AC15 Recall: Timed Reminder Recall

If the Attendant Forward No Answer feature is activated and the attendant fails to answer, the attendant is forced into Busy Position and the call goes to the first idle attendant or is put into the attendant queue. If the conditions are also satisfied to put the customer in Night Service and the original call is an external call, the AC15 recall is directed to the Night DN.

Attendant Recall

If an attendant recall is affected through the Attendant Recall key on a proprietary telephone, or through a switchhook flash on an analog (500/2500 type) telephone, the destination side on the console is not dropped before the call is routed to the night DN.

Camp-On to a Set in Ringback or Dialing

Camp-on recall takes precedence over the Attendant Forward No Answer recall. However, if during the recall the customer goes into Night Service and the recall is not answered by the night DN, the call is disconnected according to the Attendant No Answer feature processing.

DPNSS1 Diversion

If an incoming call is handled for Network Attendant Services routing towards DPNSS1, no diversion signaling is sent back to the calling party.

Multi-party Operations - Recovery of Misoperation during Call Transfer

Multi-Party Operations – Recovery of Misoperation During Call Transfer takes precedence over NFNA and NFNS for DID/DOD/CO calls.

When a DID/DOD/CO call is transferred from one station to another station on the same node, Ring Again No Answer has priority over NFNA and NFNS.

Night Forward No Answer

Call Forward No Answer has priority over Night Forward No Answer and AFNA on the Attendant Overflow Position.

Night Service Enhancements

Any call which has been presented to the Attendant Overflow Position cannot be not be removed from the set and requeued by pressing the Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Position Busy with Call on Hold

If an attendant with a call on hold does not answer an Attendant Forward No Answer call within a customer-defined time, the console is not placed in Position Busy.

Recall to Same Attendant

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call is routed to the first available idle attendant.

Switchhook Flash

If a switchhook flash is performed on an analog (500/2500 type) telephone, the AFNA timing stops to allow for a valid disconnection. If a valid disconnection is not affected, the AFNA timing cycle begins again.

Feature packaging

Attendant Forward No Answer (AFNA) is package 134; however, this package is mutually exclusive with Attendant Alternate Answering (AAA) package 174.

Within a networking environment, Network Attendant Service (NAS) package 159 is required.

Feature implementation

LD 15 – Modify data for each customer member to be configured.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT	Attendant Console options.
...		
- OPT	(DNCA) DNCS	If DNCA is entered, all DID/CO or DOD calls are disconnected after the number of ring cycles defined by the response to the NFNA prompt while the system is in Night Service. If DNCS is entered, outgoing CO/DOD calls or incoming CO/DID calls in the answered state, and waiting on a set are disconnected after the number of seconds defined in response to the NFNS prompt expires.
...		
- AFNT	(0)-2-126	Attendant Forward No Answer Timer. The number of seconds in two-second intervals that the call is presented to the attendant before Attendant Forward No Answer is attempted. Odd entries are rounded down to the next valid entry. If 0 is entered, the call is not forwarded.
- AFBT	(0)-2-x	Attendant Forward Buzz Tone, where: x = the value defined for AFNT. The number of seconds in two-second intervals that the attendant is buzzed at full volume before the Attendant Forward No Answer timer is reached. Odd entries are rounded down to the next valid entry. If 0 is entered, the original volume is in effect.
TYPE:	TIM	Timers.
...		

- NFNA	(0)-63	Night Forward No Answer ring cycles (prompted if OPT = DNCA). The number of times a DID/DOD and CO trunk call will ring a set before being disconnected during Night Service. A default value of 0 causes the call not to be disconnected.
- NFNS	(0)-504	Night Forward No Answer in seconds (prompted if OPT = DNCS). If a value is entered for this prompt, all outgoing CO/DOD trunk calls in a waiting state, and all incoming CO/DID calls in the answered state will be disconnected after the time in seconds expires as entered in response to this prompt. The entered value must be a multiple of eight. A default value of 0 causes the call not to be disconnected.

Feature operation

No specific operating procedures are required to use this feature.

Attendant Incoming Call Indicators

Contents

This section contains information on the following topics:

Feature description 323
Operating parameters 324
Feature interactions 324
Feature packaging 325
Feature implementation 325
Feature operation 326

Feature description

Attendant Consoles can be equipped with up to 20 Incoming Call Indicator (ICI) key/lamp pairs to identify the type of calls being presented and the call status for each particular call type. The customer can specify which incoming call types are to be assigned a separate ICI key. Possible call types include, but are not limited to, the following:

- Trunk calls (such as FX, WATS, and TIE)
- Listed Directory Number (LDN) calls
- Dial zero calls
- Fully restricted dial zero calls
- Automatic Timed Reminder recalls
- Attendant Interpositional calls

- Attendant Intercept calls
- Call Forward Busy calls
- Call Forward No Answer calls

Three lamp states are associated with each Incoming Call Indicator key (See Table 16).

Table 16
Key lamp states associated with each Incoming Call Indicator key

Lamp state	Status of call type
Dark	No calls of this type are waiting.
Flashing	One call of this type is waiting in queue.
Steadily lit	Two or more calls of this type are queued, or one call has been waiting longer than 20 seconds.

Operating parameters

The ICI feature applies to Attendant Consoles only.

The number of ICI keys to be assigned (10 or 20) is defined in the Customer Data block. The default is ten.

No more than 20 ICI key/lamp pairs can be assigned to an Attendant Console. The assignment of call types to ICI key/lamp pairs is flexible. All Attendant Consoles in the customer group will have the same ICI key assignments.

Feature interactions

Attendant Call Selection
Call Waiting

The ICI feature is used with the Attendant Call Selection and Call Waiting features to recognize, answer, and process incoming calls.

DPNSS1 Night Service

When a Night Service call is diverted to an attendant, the Incoming Call Indicator is the number of the incoming route. This is the same as for a NAS MCDN call routed to an attendant.

ISDN Semi Permanent Connections for Australia

Calls using an ISPC link are always presented as calls over TIE trunks.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Assign ICI keys for Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console Options.
CUST		Customer number
	0-99	For Large Systems
	0-31	For Small Systems and Succession 1000 systems
- OPT	(IC1) IC2	10 or 20 Incoming Call Indicators.
- ICI	0-19 CAx	Station category number. x = category number 1 through 7.
	0-19 CFB	Call Forward Busy.
	0-19 CFN	Call Forward No Answer.
	0-19 DF0	Dial 0 fully restricted.
	0-19 DL0	Dial 0 (attendant).
	0-19 IAT	Inter-attendant call.
	0-19 INT	Call intercept.
	0-19 LCT	Line Lockout Intercept.
	0-19 LD0-3	Listed Directory Number (0 through 3).
	0-19 MWC	Attendant Message Center.
	0-19 RLL	Recall.
	0-19 xxx	Route number.

Feature operation

No specific operating procedures are required to use this feature.

Attendant Interpositional Transfer

Contents

This section contains information on the following topics:

Feature description 327
Operating parameters 327
Feature interactions 328
Feature packaging 328
Feature implementation 329
Feature operation 329

Feature description

Attendant Interpositional Transfer enables an attendant to call or transfer a call to another attendant in a multiple console group, even when the destination Attendant Console is busy.

When transferring a call to another attendant whose console is idle, the interpositional call is presented immediately. If the called attendant is busy, the calling attendant hears a busy tone. The attendant then presses the Release key and the transferred call will be the next call presented to the called Attendant Console.

Operating parameters

A call can be transferred to an Attendant Console in the Position Busy state; however, the called console does not receive any audible signal. A Call Waiting indication appears on the console display.

Feature interactions

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Attendant Overflow Position feature is supported in a UDP DPNSS1 network. An attendant can call or transfer a call to another attendant in a multiple-console group, even when the destination Attendant Console is busy.

Network Attendant Service

An attendant is not able to call a specific attendant on another node by dialing the attendant DN followed by the attendant number. The attendant dials the NARS or CDP or LDN number the same as a telephone dials to reach the attendants at another node.

Night Service Enhancements

The requeuing of interpositional calls is not allowed. Night Service enhancements do not apply to interpositional calls, which remain on the console until answered.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Add/change an Interpositional Call Incoming Call Indicator (ICI) key on Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ICI	0-19 IAT	Add an Inter-attendant Call ICI to all consoles.

Feature operation

To transfer a call to a busy attendant (Attendant Console):

- Press **Rls**. Your call will be the next call presented to the busy attendant.

To transfer a call to an Attendant Console in Position Busy mode:

- Dial the Interpositional access code (0) and the desired attendant position number. You receive a busy tone. Press **Rls**.

To answer a call transferred to an Attendant Console in Position Busy mode, follow these steps:

1. The Call Waiting indicator lights; there are no audible tones. Press the **Position Busy** key to take the console out of Position Busy mode.
2. The call is presented to the Loop key and you receive an audible tone. Press the **Loop** key.

Attendant Lockout

Contents

This section contains information on the following topics:

Feature description 331
Operating parameters 331
Feature interactions 332
Feature packaging 332
Feature implementation 332
Feature operation 332

Feature description

Attendant Lockout restricts the attendant from entering an established connection completed through and held on the console. Attendant Lockout does not come into effect until the call has been answered.

The attendant can re-enter the call if the source party is a station telephone. Attendant Lockout occurs only if the source party is an external number (trunk), and the destination party is a telephone.

Operating parameters

Busy Verify and Barge-In allow the attendant to override the Attendant Lockout feature.

Feature interactions

Attendant Recall

If one of the stations activates Attendant Recall, the attendant is allowed to re-enter the connection.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Allow/deny Lockout for Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(LOD) LOA	(Deny) allow attendant lockout.

Feature operation

No specific operating procedures are required to use this feature.

Attendant Overflow Position

Contents

This section contains information on the following topics:

Feature description 333
Operating parameters 334
Feature interactions 336
Feature packaging 341
Feature implementation 341
Feature operation 342

Feature description

Attendant Overflow Position (AOP) allows certain types of calls to be automatically rerouted to a specified idle Directory Number (AOP DN) when calls waiting to be answered have exceeded a defined threshold, or an attendant is in the Position Busy state, but the system is not in Night Service.

When a call that can be rerouted has been waiting longer than the customer-defined Attendant Queue Timing Threshold (0-255 seconds), it is rerouted to the AOP DN. Calls that can be rerouted to the AOP DN are trunk calls, internal calls and Call Forward Busy, or Call Forward No Answer calls directed to the attendant.

Attendant calls that cannot be rerouted are transfer calls, intercept calls, parked call recalls, automatic or manual recalls, and attendant interposition calls. These calls will not be answered until an attendant becomes available.

When the last Attendant Console is put into Position Busy or disabled, the system does not go into Night Service if an AOP DN is available. In this case, calls that can be rerouted will be forwarded to the AOP DN. Ineligible calls remain unanswered until the system is put in Night Service or one of the consoles deactivates Position Busy.

Operating parameters

An AOP DN can be a single-appearance, multiple-appearance single-call, or multiple-appearance multiple-call DN. If it is a Multiple Appearance DN, a proprietary telephone can busy out the AOP DN for all appearances.

An analog (500/2500 type) telephone can have an AOP DN. It does not have the ability to busy out the AOP DN and continue to receive calls. If it is a requirement that the analog (500/2500 type) telephone have an AOP DN, the AOP DN must also appear on a proprietary telephone to create a mix of telephones, which negates privacy.

In order to properly identify and greet attendant overflow calls, it is best to have the AOP DN appear on a proprietary telephone's secondary DN.

Proprietary telephones specified as Attendant Overflow Positions can prevent calls from being rerouted by the Attendant Overflow feature. To prevent attendant overflow calls, press the Attendant Overflow Position Busy (AOP Busy) key/lamp pair on the telephone. Activating this key will busy out all appearances of the AOP for either Single Call Ringing or Multiple Call Ringing arrangements. Overflow calls will remain in the attendant queue. Normal incoming calls to the AOP telephone will not be affected.

The following requirements apply to the activation/deactivation of the AOP Busy key:

- A telephone with an AOP Busy key must have an appearance of the AOP DN in order for the key to work.
- Any AOP DN that has an AOP Busy key can activate or deactivate the AOP feature. If the AOP Busy key is activated at one appearance of the AOP DN, attendant calls are not rerouted to any appearance of the AOP DN.
- Activation or deactivation of the AOP Busy key does not affect any call already rerouted to the AOP DN.

- If all consoles are in Position Busy and the system is not in Night Service when an AOP Busy key is activated, the system goes into Night Service.
- If the system is in Night Service when the AOP Busy key is deactivated, the system remains in Night Service.
- Activation or deactivation of the AOP Busy key does not affect the Position Busy status of the Attendant Console. If all Attendant Consoles are in Position Busy and the AOP Busy key is activated, the system goes into Night Service.
- The status of the AOP Busy key remains unchanged through a system initialization but is deactivated if a system reload occurs.

The CAS to AOP Interworking feature allows both Centralized Attendant Service-Main (CASM), or Centralized Attendant Service-Remote (CASR), and Attendant Overflow Position packages to be configured and co-exist in a network. In an environment where both packages are configured, CAS takes precedence over AOP.

Each customer may have only one AOP DN. The AOP DN cannot be a private line DN, a trunk DN, a Control DN, a BRI DN, or a SPRE code.

There are no special ringing cadences or lamp operations to indicate that an incoming call to the AOP DN is an Attendant Overflow Position call. It is recommended that the AOP DN be used only for Attendant Overflow Position calls enabling calls to be answered appropriately.

If the AOP DN is busy, calls remain in the attendant queue and are not rerouted through the Attendant Overflow Position feature until the DN is free to receive the next call.

Calls will not be rerouted to the Attendant Overflow Position DN when

- Calls are on an Integrated Services Digital Network (ISDN) or Electronic Switched Network (ESN) network.
- All appearances of the AOP DN are busy.
- The AOP DN is in the Call Forward All Calls mode.
- The call is an interposition call from an attendant.
- The call has been redirected to the attendant by the Call Transfer or Attendant Recall features.

- The call is an intercept call to the attendants.
- The system is in the Power Fail Transfer mode.
- All appearances of the AOP DN have the Make Set Busy feature activated.
- Any appearance of the AOP DN has activated Attendant Overflow Position Busy (AOP Busy).
- An analog (500/2500 type) telephone appearance of the AOP DN goes idle and a Call Waiting call is queued for the telephone. The Call Waiting call rings the telephone and AOP calls are not rerouted to the telephone.
- The AOP DN goes idle with a Camp-On call queued for the telephone. The Camp-On call rings the telephone and AOP calls are not rerouted to the telephone.
- The rerouting of the call violates the access restrictions or Class of Service restrictions on the AOP DN telephone. For example, if the AOP DN is FR2, an external Public Exchange network call will not be rerouted to the AOP DN because it is prohibited by the telephone access restrictions.
- The system is in Night Service.

Feature interactions

AC15 Recall: Timed Reminder Recall

AC15 recalls are not routed to the Attendant Overflow Position. They are directed to the first idle attendant or put in the attendant queue.

Attendant

The Calls Waiting indicator on the Attendant Console is updated when a call is rerouted to the AOP DN.

Attendant Overflow Position Busy

If the telephone with Attendant Overflow Position (AOP) DN has an Attendant Overflow Position Busy (AOP Busy) key activated, calls will not overflow to any appearance of the AOP DN.

Attendant Recall

An Attendant Overflow Position call answered at an AOP DN may be recalled to the attendant using the Attendant Recall capability (ARC key).

Attendant Timed Recall
Automatic Timed Reminders

After an attendant call has been rerouted using the AOP feature, there is no automatic timed recall to the attendant or any other DN.

Automatic Call Distribution

Externally marked trunks will overflow to an Automatic Call Distribution (ACD) DN. The ACD DN may only be an ACD agent configured as a virtual Voice Mail System agent (for example, Call Pilot).

Automatic Wake Up

Automatic Wake Up recalls are not redirected to a customer-defined Attendant Overflow Position DN. Failed wake up calls stay in the attendant queue or ring indefinitely on the console.

Call Forward All Calls

If the telephone assigned an Attendant Overflow DN has activated the Call Forward All Calls feature, overflow calls are not rerouted to the telephone. If an analog (500/2500 type) telephone is forwarded, AOP is canceled.

Call Forward, Internal Calls

If Attendant Overflow redirects an internal call to a telephone that is Internal Call Forward active, the call will remain in the attendant queue, and will not receive Internal CFW treatment.

Call Forward No Answer

A call rerouted through Attendant Overflow Position will Call Forward to the forwarding DN only if it is the Prime DN or a single appearance DN on that telephone.

Call Pickup

An Attendant Overflow Position Call presented to the AOP DN can be picked up by any station belonging to the same Call Pickup Group.

Conference

An Attendant Overflow Position call answered on an AOP DN may be conferenced with another DN.

Departmental Listed Directory Number

Listed Directory Number calls that have been waiting in the queue longer than the specified threshold period will be routed to the Attendant Overflow Position.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Attendant Overflow Position feature is supported on a UDP DPNSS1 network. If an incoming DPNSS1 UDP call is queued to the attendant, and if the call is not answered within a predefined period of time, the call can be redirected to the Attendant Overflow DN.

Flexible Attendant Call Waiting Thresholds

The Attendant Overflow Position is not counted as an active attendant.

Flexible Line Lockout

A call intercepted to the attendant due to Flexible Line Lockout receives Attendant Overflow Position (AOP) treatment if the feature package is equipped and the AOP Directory Number (DN) is defined.

Group Hunt

A PLDN cannot be configured as an Attendant Overflow DN (AODN).

Line Lockout

If a telephone with an AOP DN is in Line Lockout, it still receives AOP calls.

Make Set Busy

If a telephone that is the only idle AOP DN has MSB activated, calls will not overflow.

If the AOP DN is a multiple appearance DN, the MSB key should be added to all telephones with an AOP DN.

If MSB is activated in a Multiple Call Ringing arrangement, the telephone appears busy. All other appearances of the AOP DN will still receive calls. This allows the user to leave the telephone and prevent callers from overflowing and receiving ringback with no answer.

If the AOP DN is a Multiple Appearance, Single Call arrangement and MSB is activated, the AOP DN of that telephone will flash, but the telephone will not ring (the call can still be answered from that appearance).

Manual Line Service

When Attendant Overflow Position (AOP) is defined, Manual Line Service follows the AOP directions.

Meridian Hospitality Voice Services

Attendant Overflow Position (AOP) allows unanswered calls to the attendant to be forwarded to a customer-defined Directory Number (DN) after a defined time. A call can also be overflowed if all the attendants are in Position Busy State. With AOP equipped , overflowed calls can be directed to Call Pilot. The AOP DN must be defined as an Automatic Call Distribution (ACD) Directory Number (DN), and the ACD DN must have an ACD agent assigned as a virtual VMS agent.

Multiple Appearance Directory Number

A multiple appearance, multiple call AOP DN allows as many overflow calls to be in progress as there are appearances of the DN. A multiple appearance, single call AOP DN allows only one overflow call at a time.

Night Key for Direct Inward Dialing Digit Manipulation

When the last attendant activates the POS BUSY key, the system does not go into Night Service if an Attendant Overflow Position Directory Number (DN) is available.

Night Service

A call rerouted through the Attendant Overflow Position feature is not redirected to the Night DN if the system is subsequently put into Night Service. When all Attendant Consoles are in Position Busy, the system will not go into Night Service until the AOP Busy key is activated.

Deactivating the AOP Busy key after the system has been placed in Night Service does not affect the Night Service feature.

Night Service Enhancements

If a call with a ringing party on the destination side is presented at the last-active Attendant Console, and there is an active Attendant Overflow Position, the ringing destination will be disconnected when the call is requeued. Likewise, if the call is a Call Waiting recall, Call Waiting will be canceled.

Night Service Enhancements/Network Attendant Service (NAS)

The routing configuration for NAS will apply during Night Service. External calls and recalls may be queued to a remote Night DN, if defined. Internal calls and internal recalls queued during Day Service will be dropped, if the Night DN has been defined on a remote node.

Recall to Same Attendant

Recalls and inter-attendant calls are not routed to the Attendant Overflow Position.

Ring Again

If Ring Again is activated against the AOP DN, notification is given to the originator when the telephone becomes idle. An AOP call, however, takes precedence over Ring Again notification on the AOP DN when the AOP DN becomes free.

Traffic Measurement

Traffic measurements are provided for the Attendant Overflow feature in Traffic Report TFC005. A count of the number of attendant calls rerouted through the feature is printed.

Feature packaging

Attendant Overflow Position (AOP) package 56 has no feature package dependencies. Attendant Overflow Position and Centralized Attendant Service are, however, mutually exclusive.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Assign/change an Attendant Overflow Position DN and queue threshold timing.

2 LD 11 – Add/change an AOP DN and AOP Busy key.

3 LD 10 – Add/change an Attendant Overflow Position DN on an analog (500/2500 type) telephone.

LD 15 – Assign/change an Attendant Overflow Position DN and queue threshold timing.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- AQTT	0-(30)-255	Attendant queue timing threshold (AQTT).
- AODN	xxx…x	DN where calls are to be overflowed when they have been in queue the time specified for AQTT.

LD 11 – Add/change an AOP DN and AOP Busy key.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx yyy…y	Attendant Overflow Position DN. xx = key number. yyy…y = DN.
KEY	xx OVB	Attendant Overflow Position Busy key.

LD 10 – Add/change an Attendant Overflow Position DN on an analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DN	yyy…y	Attendant Overflow Position DN.

Feature operation

Attendant Overflow Position calls will be rerouted to all appearances of the AOP DN as long as the following conditions are met:

- The system is not in Night Service.

- The Attendant Overflow key (any AOP DN appearance) is not activated.
- At least one appearance of the AOP DN is on a telephone that does not have Make Set Busy activated.

To prevent attendant overflow calls from being rerouted to the AOP DN, do any of the following:

- Activate the Attendant Overflow Position Busy key.
- Activate the Make Set Busy key on all telephones with an appearance of the AOP DN.
- Place the system in Night Service.

To prevent attendant overflow calls from being rerouted to a single telephone with an appearance of the AOP DN (but not others):

- Activate Make Set Busy or
- Activate Call Forward All Calls (analog (500/2500 type) telephone).

Attendant Position Busy

Contents

This section contains information on the following topics:

Feature description 345
Operating parameters 345
Feature interactions 346
Feature packaging 347
Feature implementation 347
Feature operation 347

Feature description

If multiple consoles are defined for a customer, an attendant can remove a console from service by pressing the Position Busy key. Incoming calls are then directed to other consoles in the customer group.

Operating parameters

Position Busy applies to Attendant Consoles only.

Feature interactions

Attendant Administration

If a console in the Attendant Administration mode is idle for more than 20 minutes, it automatically reverts to Position Busy. If the system is initialized or reloaded while the console is in Attendant Administration mode, Attendant Administration is aborted and the console is placed in Position Busy.

Attendant Supervisory Console

Activation of the Position Busy key on a Supervisory console puts the console in the supervisory mode.

Departmental Listed Directory Number

If all Attendant Consoles in an LDN group are in a Position Busy state, calls to that LDN will not be automatically presented to any Attendant Console in the customer group. Other attendants may only answer those LDN calls if the LDN has been assigned to an ICI key.

End-to-End Signaling

Attendant Position Busy works together with Attendant End-to-End Signaling (AEES). However, do not press this feature key while using AEES, or the Dual-tone Multifrequency (DTMF) code signals may be blocked.

Night Service

When the last console operator activates the Position Busy key or the Night key, Night Service is put into effect. Incoming calls receive the customer-specified night treatment.

When all attendants activate the Position Busy key, Night Service is in effect unless the Attendant Overflow Position (AOP) feature is equipped. If AOP is equipped, the Night key must be pressed to invoke Night Service. A call that is rerouted due to AOP is not redirected to the Night DN if the system is subsequently put into Night Service.

Night Service Enhancements

Any call that has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will be removed only if the Attendant Forward No Answer feature is active and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Recall to Same Attendant

If an Attendant Console is in maintenance or Position Busy when a Recall to Same Attendant call is recalled to it, the recall is presented to the first available idle attendant. If an attendant goes into Position Busy with a Return to Same Attendant call in Call Waiting, the waiting call is presented to the first available attendant.

Series Call

If the attendant activates Position Busy while a Series Call is active, the recall occurs to the next available attendant.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

In a multi-console environment, press the **Position Busy** key on an Attendant Console to remove it from service.

Attendant Recall

Contents

This section contains information on the following topics:

Feature description 349
Operating parameters 350
Feature interactions 350
Feature packaging 353
Feature implementation 353
Feature operation 355

Feature description

Attendant Recall allows a user to call the attendant directly during an established call by pressing a single key. A three-way connection is established among the user, the attendant, and the third party.

To activate this feature, a separate Attendant Recall key/lamp pair must be equipped on proprietary telephones. A softkey must be programmed on the M3000 Touchphone for this feature.

On single-line telephones, a user can recall the attendant during an established call by flashing the switchhook. Attendant Recall is automatic if a Transfer Denied Class of Service (XFD) is specified for the telephone. If a Transfer Allowed Class of Service (XFA) is specified, the user hears a special dial tone following the switchhook flash, and then dials zero (0) to recall the attendant. After a switchhook flash has been used to recall the attendant, it is not possible to return to a two-party connection before the attendant answers.

Operating parameters

In order for the Overflow Position Busy (OVB) key to work, the telephone must have an AOP DN configured.

Feature interactions

Attendant Alternative Answering

Under Attendant Recall conditions, the initiator of the recall rings the destination side of the console, and the third party becomes the source. The AAA timer is applied to the source party. If the AAA timer expires, the destination is dropped, and the source is forwarded to the AAA DN. If the source party disconnects before the destination party, the AAA timer is restarted on the destination party still buzzing the attendant through the ARC key. The AAA timer is dropped if both parties disconnect.

Attendant Forward No Answer

If an attendant recall is affected through the Attendant Recall key on a proprietary telephone, or through a switchhook flash on an analog (500/2500 type) telephone, the destination side on the console is not dropped before the call is routed to the night DN.

Attendant Lockout

If one of the stations activates Attendant Recall, the attendant is allowed to reenter the connection.

Attendant Overflow Position

An Attendant Overflow Position call answered at an AOP DN may be recalled to the attendant using the Attendant Recall capability (ARC key).

Attendant Secrecy

Attendant Secrecy does not apply on an attendant recall or when the attendant reenters a call held on a Loop key. The Exclude Source and Destination keys are used in these cases.

Attendant Splitting

After the attendant and the two parties have been connected, the attendant can use the Attendant Splitting feature to communicate separately with either party.

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Attendant Recall (ARC) key is ignored.

Call Party Name Display

Attendant Recall using the Attendant Recall key or a switchhook flash results in both source and destination information being displayed. No redirection reason is displayed, however. In this type of recall, the party that pressed the Attendant Recall key or switchhook is the destination party.

Attendant Recall using Call Transfer or Conference displays the recalling party's DN and CPND information on the attendant's source line. No redirection reason is displayed. If the recall is done with the Transfer key the third party's DN and CPND information are displayed on the source line when the transfer is complete.

Directory Number Delayed Ringing

If a dialed set has Directory Number Delayed Ringing (DNDR) defined, and an attendant re-extends a call without releasing it, the DNDR timing is not reset. If the value of the recall timer is less than that of the DNDR timer, the call is recalled to the attendant before audible notification begins.

Direct Inward Dialing Call Forward No Answer Timer

The Direct Inward Dialing Call Forward No Answer Timer does not apply to an answered DID call that is extended to an unanswered station by the attendant – the call is recalled to the attendant using the Attendant Recall feature.

In-Band Automatic Number Identification

If an Automatic Call Distribution Agent is active on an IANI call and activates the Attendant Recall (ARC) key to call the attendant, the agent's display shows the attendant number when the attendant answers the call. The ANI number reappears when the attendant releases.

Incoming Call Indicator Enhancement

If an RDI-intercepted call that is extended by the attendant to the destination party having RDI Class of Service is either transferred back or recalled to the attendant, then the attendant recall ICI lights up and not the RDI-intercept ICI.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

When a call from another node is recalled to the Intercept Computer (ICP) position attendant, it is presented on the ICP terminal.

Multi-Party Operations

Users of analog (500/2500 type) telephones can perform an attendant recall during a two-party connection by performing a switchhook flash and then dialing the attendant DN.

Ring Again on No Answer

A set that is recalling the attendant cannot apply Ring Again on No Answer.

Secrecy Enhancement

The source and destination parties cannot be joined together on the attendants conference bridge if Attendant Break-In with Secrecy is active. This is consistent with the existing Break-In feature.

Slow Answer Recall for Transferred External Trunks

Slow Answer Recall Modification (SLAM) has an interaction after the attendant answers the recall. If SLAM is configured, the target set is disconnected after the attendant answers the recall. If SLAM is not configured, the target set rings until the attendant releases it.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Add/change a Recall Incoming Call Indicator (ICI) key on Attendant Consoles.
2. LD 10 – Implement Attendant Recall for analog (500/2500 type) telephones.
3. LD 11 – Add/change an Attendant Recall key for proprietary telephones.

LD 15 – Add/change a Recall Incoming Call Indicator (ICI) key on Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ICI	xx RLL	Add a Recall ICI to all consoles.

LD 10 – Implement Attendant Recall for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD), XFA	(Deny) allow call transfer, which allows automatic Attendant Recall.

LD 11 – Add/change an Attendant Recall key for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx ARC	Add an Attendant Recall key (the M3000 must use key 33). xx = key number.

Feature operation

To contact an attendant during a call (Meridian 1 proprietary telephone) , follow these steps:

1 Press **Att Recall**.

2 Stay on the line until the attendant answers.

3 When you hang up, the other party remains connected to the attendant.

To contact an attendant during a call (analog (500/2500 type) telephone with Transfer Allowed Class of Service), follow these steps:

1 Flash the switchhook (you hear a special dial tone).

2 Dial zero (0).

3 When you hang up, the other party remains connected to the attendant.

To contact an attendant during a call (analog (500/2500 type) telephone with Transfer Denied Class of Service), follow these steps:

1 Flash the switchhook (the attendant is automatically dialed).

2 When you hang up, the other party remains connected to the attendant.

Attendant Recall with Splitting

Contents

This section contains information on the following topics:

Feature description 357
Operating parameters 358
Feature interactions 358
Feature packaging 360
Feature implementation 360
Feature operation 360

Feature description

The Attendant Recall with Splitting feature provides an enhancement to the operation of the Attendant Console with the following features:

- Attendant Recall
- Call Transfer

This feature allows calls transferred to the attendant by the above features to be presented on the console loop with both the transferring and transferred parties on the console loop, with the transferred party automatically excluded if OPT in LD 15 is set to either SYA (Secrecy Allowed) or EHS (Enhanced Secrecy). Upon answering the call, the attendant then assumes control over both the transferred and transferring parties. The operation will also allow the transferring party to have control over the call as long as the call has not been answered by the attendant (example, the transferring party will be able to cancel the call transfer and return to the transferred party).

It is important to note that this enhancement applies to calls transferred to the attendant using Attendant Recall and Call Transfer only. Calls transferred to the attendant through operation of the Conference key on proprietary telephones, or through the operation of the Interpositional Call Transfers, do not receive splitting.

Operating parameters

This feature applies only to calls which arrive at the attendant by way of Attendant Recall or Call Transfer.

This feature will not function across a network.

This feature requires OPT in LD 15 (Customer Data Block) be set to either SYA or EHS.

Feature interactions

Attendant Secrecy

Secrecy Allowed (SYA)

If Secrecy is allowed at the Attendant Console, a two-party connection will be made only when the attendant answers the call. The attendant can converse privately with either the source or the destination side (Splitting) until the Loop key is pressed and a three-party connection is reestablished.

Secrecy Denied (SYD)

If Secrecy is denied at the Attendant Console, a three-way connection will be established between the transferring party, transferred party, and the attendant when the attendant answers the call.

Enhanced Secrecy (EHS)

Same as SYA except that a warning tone is included as part of all conversations involving the attendant and two or more parties to indicate that privacy has been interrupted.

Automatic Call Distribution (ACD)

A recall from an ACD DN to the attendant console will also activate the Attendant Recall with Splitting feature. The call is treated as if it had come from a normal internal DN instead of an ACD agent. The operation is described in *"Normal Operation" on page 360*

Automatic Hold

This feature does not have precedence over Attendant Recall (that is, automatic hold cannot be activated until the attendant answers the recall presented on the console). However, it can be activated even before the attendant answers a call transferred to the console.

Call Detail Recording (CDR) on Multiple Call Transfer

With PPM

Whenever a PPM call is transferred, the pulses accumulated against the current station that is responsible for this segment of the call are added to its terminal meter and a CDR X (an S for the first time) record is printed. When the call is eventually terminated, a CDR E record is printed.

Without PPM

The type and number of CDR records printed will be the same as the case for outgoing PPM call. The only difference is that no accumulated pulses will be included as part of the CDR messages.

Intercept Computer Dial from Directory

If a set transfers a call to the attendant, or a Meridian 1 proprietary telephone presses the Attendant Recall (ARC) key and the transferring party has not yet completed the transfer before the attendant answers, it is not possible to dial from the Intercept Computer (since the transferred party is connected to SRC, and the transferring party is connected to DEST).

Call Party Name Display

For the M1250 and M2250 Attendant Console, M2317, M3000 digital sets, and Meridian Modular sets the appropriate DN and calling party's name will be correctly shown on the digit display when the attendant presses either the Exclude Source or the Exclude Destination key.

Multi-Party Operations

The Multi-Party Operations (MPO) feature introduces a new Class of Service; Three Parties Service Allowed (TSA), for analog (500/2500 type) telephones. It allows certain keys on these sets to be programmed for conference, toggle between sets, and disconnect. However, the toggle function will be disabled if a call is transferred to the attendant because of the Attendant Recall with Splitting feature.

Slow Answer Recall Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Transfer Restricted

This feature ignores the use of switchhook flash on analog (500/2500 type) telephones and as a result call transfer, conference, and attendant recall (with or without splitting) will not be allowed on a set basis.

Feature packaging

Attendant Recall with Splitting requires International Supplementary Features (SUPP) package 131.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

Normal Operation

The following events describe the normal operation whenever a call is transferred to the attendant using the Call Transfer feature at an analog (500/2500 type) telephone, or using the Call Transfer/Attendant Recall feature at a proprietary telephone, or using the operation of a register recall at an analog (500/2500 type) telephone with Transfer Denied Class of Service.

If an Attendant Console is idle, then the call will be presented to the console as follows:

- The Loop indicator stays off.
- The Recall Incoming Call Identification (ICI) indicator is turned on (with other ICI indicators associated with waiting calls of other types).
- The Source indicator for the loop on which the call is presented is turned on.
- The Destination indicator for the same loop flashes at 120 ipm.
- The console buzzes.
- The Attendant Console digit display indicates the DN (or name if CPND package is enabled) of the transferring party.
- The transferring party receives ringback tone.
- The transferred party is put on hold.

If no Attendant Console is idle when the transferring party dials the attendant access code, then the call is placed in the attendant queue, and the transferring party receives ringback tone. When this call moves to the top of the queue and an Attendant Console becomes idle, then the call will be presented to the console as described in the previous paragraph.

The attendant can then answer this call by pressing the Loop key, or by pressing the Recall ICI key. When the call is answered, the following occurs:

- The Loop indicator is turned on.
- The Recall ICI indicator stays on, and all other ICI indicators are turned off.
- The controlling party is presented at the Attendant Console as a destination, and the Destination indicator stays on steadily.
- The transferred party is presented at the console as a source, and the Source indicator remains on.
- The source (the transferred party) is automatically excluded from the connection, and the Exclude Source indicator is turned on.

- The destination (the transferring party) is connected to the attendant.

  The previous two events only occur when the SYA or EHS option is allowed. If SYD is defined in the Customer Data Block, a three party conference will be set up instead.

- If the call is transferred from a proprietary telephone, the proprietary telephone's Attendant Recall indicator or the Call Transfer indicator is turned off.

If the call is transferred from an M2317 or M3000 set, then the screen on the corresponding set will go to the established state.

The attendant then assumes control over both the source and the destination as if both parties have been dialed by the attendant.

However, the transferring party can either return to the transferred party or complete the transfer operation any time before the attendant answers the call (that is, while the call is presented to the console, or placed in the attendant queue).

Cancel Call Transfer

The station user can return to the original party (the transferred party), before the attendant answers the call, as follows:

- The analog (500/2500 type) telephone user: By operating the register recall again, which causes the call to revert back to a two-party call, and the call to the attendant to be canceled.
- The proprietary telephone user: By pressing the DN key (DN indicator flashes at 120 ipm), which causes the call to revert back to a two-party call, the call to the attendant to be canceled, the DN indicator to stop flashing and stay on steadily, and the Call Transfer indicator (or the attendant Recall indicator) to turn off.
- By pressing the Release key or going on-hook, which causes the call to revert back to a two-party call and to be put on hold, the call to the attendant to be canceled, the Call Transfer indicator (or the attendant Recall indicator) to turn off, and the DN indicator to flash at 120 ipm.

- The M2317 set user: By pressing the DN key (DN indicator flashes at 120 ipm), which causes the call to revert back to a two-party call, the call to the attendant to be canceled, the DN indicator to turn on steadily.

  By pressing the Release key or going on-hook, which causes the call to revert back to a two-party call and to be put on hold, the call to the attendant to be canceled, (the attendant Recall indicator to turn off), and the DN indicator to flash at 120 ipm.

- The M3000 set user: By pressing the DN key (DN indicator flashes at 120 ipm), which causes the call to revert back to a two-party call, the call to the attendant to be canceled, the DN indicator to turn on steadily.

  By pressing the Release key or going on-hook, which causes the call to revert back to a two-party call and to be put on hold, the call to the attendant to be canceled, and the DN indicator to flash at 120 ipm.

  Note: Pressing the DN key or operating the recall after the attendant answers the recall will be ignored.

Complete Call Transfer

While waiting for the attendant to answer the recall (ringback tone is received), the station user can complete the call transfer to the attendant as follows:

- The analog (500/2500 type) telephone user: By going on-hook, which causes the analog (500/2500 type) telephone to become idle and the attendant will ring.
- The proprietary telephone user: By pressing the Call Transfer key (or the attendant Recall key), which causes the DN indicator to turn off, the Call Transfer indicator (or the attendant Recall indicator) to turn off, and the DN to become idle.
- The M2317 set user: By pressing the CONNECT soft key (or the attendant Recall key), which causes the DN indicator to turn off, (the attendant Recall indicator to turn off), and the DN to become idle.
- The M3000 set user: By pressing the JOIN PARTIES function on the touch screen, which causes the DN indicator to turn off and the DN to become idle.

If the transfer operation is completed while the call is presented to the console, then the following will occur:

- The Destination indicator turns off.
- The Source indicator stays on steadily.
- The Attendant Console digit display changes to identify the transferred party.
- The transferred party receives ringback tone.
- The Recall ICI indicator stays on steadily (with other ICI indicators associated with waiting calls of other types).
- The console continues to buzz.

If the transfer operation is completed while the recall is in the attendant queue, then the DN at which the call is transferred becomes idle, the transferred party receives ringback tone, and the call stays in the queue as a recall.

> ***Note:*** Operation is not allowed after the attendant answers the recall. The transferring party cannot drop from the call in this case until the attendant presses the Release Destination key.

Attendant Secrecy

Contents

This section contains information on the following topics:

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Operating parameters 365
Feature interactions 366
Feature packaging 367
Feature implementation 367
Feature operation 367

Feature description

Attendant Secrecy automatically prevents a voice connection between the source and destination parties of a call being extended by an attendant, until the attendant connects the two parties. This allows the attendant to converse privately with the destination party before completing the connection. Attendant Secrecy is allowed or denied on a customer basis.

Operating parameters

Attendant Secrecy is available on Attendant Consoles only.

Attendant Secrecy operates only on external calls received from an outside trunk (for example, Central Office or WATS trunks).

Attendant Secrecy is not applicable to Integrated Services Access (ISA) trunks.

Feature interactions

AC15 Recall: Timed Reminder Recall

Secrecy is not activated when AC15 recalls are presented to the attendant.

Attendant Recall

Attendant Secrecy does not apply on an attendant recall or when the attendant reenters a call held on a Loop key. The Exclude Source and Destination keys are used in these cases.

Attendant Recall with Splitting

Secrecy Allowed (SYA)

If Secrecy is allowed at the Attendant Console, a two-party connection will be made only when the attendant answers the call. The attendant can converse privately with either the source or the destination side (Splitting) until the Loop key is pressed and a three-party connection is reestablished.

Secrecy Denied (SYD)

If Secrecy is denied at the Attendant Console, a three-way connection will be established between the transferring party, transferred party, and the attendant when the attendant answers the call.

This is the Enhanced Secrecy (EHS)

Same as Secrecy Allowed except that a warning tone is included as part of all conversations involving the attendant and two or more parties to indicate that privacy has been interrupted.

Console Presentation Group Level Services

The Secrecy option specified for a customer applies to all attendants for that customer.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

If attendant secrecy is not active when the attendant attempts Executive Intrusion, the source is automatically excluded. If Enhanced Secrecy is equipped, source exclusion includes the removal of the Enhanced Secrecy warning tone when Executive Intrusion is activated.

Music

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Allow/deny Attendant Secrecy for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(SYD) SYA	(Deny) allow Attendant Secrecy.

Feature operation

No specific operating procedures are required to use this feature.

Attendant Splitting

Contents

This section contains information on the following topics:

Feature description 369
Operating parameters 369
Feature interactions 370
Feature packaging 370
Feature implementation 370
Feature operation 370

Feature description

Attendant Splitting allows the attendant to talk privately to the source or destination side of an existing connection on the console. The Exclude Source (EXCL SRC) key allows the attendant to speak privately with the destination (called) party. The Exclude Destination (EXCL DEST) key allows the attendant to speak privately with the source (calling) party.

Operating parameters

This feature is active only while the attendant is involved in the call.

Attendant Splitting applies to Attendant Consoles only.

Feature interactions

Attendant Recall

After the attendant and the two parties have been connected, the attendant can use the Attendant Splitting feature to communicate separately with either party.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

To speak privately to the source party:

1 Press **EXCL DEST**.

2 To connect yourself, the caller, and the called party, press the **lpk** key.

3 To end your connection in the call, press **Rls**.

To speak privately to the destination party:

1 Press **EXCL SCR**.

2 To connect yourself, the caller, and the called party, press the **lpk** key.

3 To end your connection in the call, press **Rls**.

Attendant Supervisory Console

Contents

This section contains information on the following topics:

Feature description 371
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Feature interactions 376
Feature packaging 377
Feature implementation 377
Feature operation 381

Feature description

The Supervisory Console feature allows one Attendant Console in a customer group to function in a supervisory capacity when put into the Position Busy state. The elements of the Supervisory Console feature allow any of the following functions.

Attendant Status Display

The supervisor, by monitoring the attendant status display, can determine how many attendant positions are in service and able to receive calls.

M1250 console – If 1 to 16 attendants are assigned within a customer group, the supervisory console can monitor their status using Trunk Group Busy keys. No add-on module is necessary.

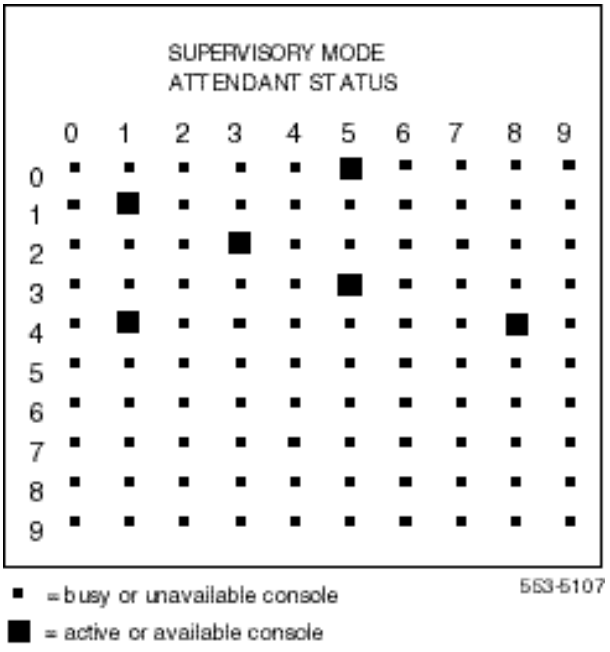
M2250 console – If 1 to 20 attendants are assigned within a customer group, the supervisory console can monitor their status using Trunk Group Busy keys. No add-on module is necessary.

When an indicator on the module associated with a particular attendant is on, the attendant is available to service calls. If the indicator is off, the attendant position is in a Position Busy state. Attendant status indicators are only operable when the supervisory console is in a supervisory mode (Position Busy key operated). When the supervisory attendant is in Position Busy, the LED associated with the supervisor fast flashes at 120 ipm.

Attendant Status using Lamp Field Array

M1250 and M2250 consoles – A supervisory console can have up to 49 status indicators when used in the Standard Busy Lamp Field mode. When using Enhanced Busy Lamp Field mode, a supervisory console can display the status of all Attendant Consoles in the customer group. Figure 5 shows an example of Supervisory monitoring in Enhanced Busy Lamp Field mode on the Busy Lamp Field/Console Graphics Module.

Figure 5
Enhanced Busy Lamp Field Supervisory mode



Visual indication of calls in queue

An attendant call queue holds incoming calls to the system that cannot be immediately answered by attendants. The supervisory console can monitor the call queue for specific types of incoming calls.

A maximum of 20 (ICI) key/lamp pairs can be assigned on an Attendant Console. Each ICI is assigned to handle a specific type of call (such as station, TIE, or dial 0) to the attendant. When a console is in the supervisory mode, the state of the lamp associated with each ICI provides a visual indication of the number of calls in the attendant queue for each ICI type. Each supervisory console ICI lamp state (dark, flash at 60 ipm, fast flash at 120 ipm, steadily ON) provides the supervisor with a visual indication of the number of calls in the queue for each call type. The ranges (calls in queue) are identified by one of three customer-specified thresholds that are set in service change programs.

Attendant Service Observation

This feature allows the supervisory attendant to monitor (listen only) calls in progress on other attendant loops without being heard. Service Observation requires the assignment of one key/lamp pair on the supervisory console flexible key strip. The key is assigned as Busy Verify through service-change programs. When the console is in Supervisory mode, the key function is Service Observation; when the console is operating as a normal attendant the key function is Busy Verify.

The observed attendant and the connected party or parties are not aware that their conversation is being monitored. The supervisor can release the connection by pressing the Release key. When the attendant is in a Service Observe mode, only the Release key is allowed as a valid input.

Supervisory assistance

An attendant can consult with, or transfer calls to, the supervisor or another attendant using the Interposition call feature. Interposition calls to the supervisor are allowed regardless of the mode of operation (Supervisory or Attendant). The supervisor can use the Interposition call feature to contact any attendant, except those in Position Busy. When the supervisor is conferring with an attendant, subsequent calls to the supervisor receive a busy indication.

If an attendant calls the supervisor who at the time is not in supervisory mode and is handling a call, the supervisory attendant interposition ICI lamp flashes at 60 ipm. As soon as the supervisor is idle, the calling attendant is connected to an idle loop on the supervisory console.

Interposition calls can be made from any attendant in the customer group to any other attendant within the customer group. Only one interposition call can be terminated on a console at a given time.

Supervisor serving as attendant

When the supervisor decides to act as an attendant, the supervisory console is removed from Position Busy. The system presents calls to the supervisory console as if it were a normal Attendant Console. The supervisory console must be idle to change states from attendant to supervisor or supervisor to attendant.

Operating parameters

The supervisory console and all Attendant Consoles (except M2250 Attendant Consoles) in the customer group must be assigned to QPC297 Attendant Console Monitor circuit packs. Their prime TN must be assigned to unit 0 and the secondary TN must be assigned to unit 1. Units 2 and 3 can be used for power; otherwise they must be left unassigned.

> ***Note:*** M2250 digital Attendant Consoles must be a minimum vintage of AD and have the Attendant Supervisory Module (ASM) installed to allow supervision.

The supervisory console must have a Digit Display (DDS).

An M1250 or M2250 console equipped with a Busy Lamp Field/Console Graphics Module (BLF/CGM) can display the status of all Attendant Consoles (up to the maximum 63) by using the Enhanced Busy Lamp Field mode. The BLF/CGM must be minimum vintage AD to provide this capability.

One supervisory console can be assigned per customer. Only one Attendant Console (1 to 63) can be assigned as a supervisory console.

The customer group must be equipped with more than one attendant.

When using the Attendant Supervisory Module (ASM), the console TN must be configured on unit 0, 4, 8, 16, and so on. The secondary TN (SETN) unit must succeed the Primary TN (1, 5, 9, 17, and so on). The ASM TN is then configured with TYPE = PWR. The PWR TN must succeed the SETN (2, 6, 10, 18, and so on).

Feature interactions

Add-on modules

Add-on modules (key/lamp strips and lamp field arrays used to display attendant status) can be used for other purposes defined by the customer when the console is in Normal mode; however if the Busy Lamp Field is assigned to display attendant status, it cannot be used for other functions during any mode of the Attendant Console.

Attendant Administration

Attendant Administration mode can be entered directly from the supervisory console from Supervisory or Normal mode by pressing the program (PRG) key. The Supervisory mode does not need to be terminated first.

Attendant Position Busy

Activation of the Position Busy key on a Supervisory console puts the console in the supervisory mode.

Controlled Class of Service, Enhanced

When the attendant is in the supervisory mode, Controlled Class of Service programming is prohibited.

Console Presentation Group Level Services

The supervisory console specified for a customer belongs to one Console Presentation Group (CPG). In the Supervisory mode, ICI indicators show only the information for ICIs in that CPG. Thresholds specified in the Customer Data Block apply only to the CPG where that console resides, and do not effect any other CPG.

Departmental Listed Directory Number

The supervisory capabilities extend to all Attendant Consoles defined within the customer group. The Attendant Console serving as supervisor should be a member of every Departmental Listed Directory Number group so that it can serve all groups when operating in the Normal mode.

End-to-End Signaling

The supervisor can operate Attendant End-to-End Signaling (AEES) if there is a call on the active loop key. An attendant in AEES mode can be monitored by the supervisor.

Multi-Tenant Service

The supervisory capabilities extend to all Attendant Consoles defined within the customer group, regardless of tenant partitioning. The Attendant Console serving as supervisor should be a member of every Call Presentation Group so that it can serve all Tenant groups when operating in the Normal mode.

Source Included when Attendant Dials

While the attendant dials the destination, the source receives intrusion tone.

Feature packaging

Supervisory Console (SUPV) package 93 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Enable/disable feature for an M1250/2250 console with a Console Graphics Module in the Standard Busy Lamp Field mode.
2. LD 15 – Enable/disable feature for an M1250/2250 console with a Console Graphics Module in the Enhanced Busy Lamp Field mode.
3. LD 12 – Enable/disable supervisory console Silent Observe.

4 LD 12 – Enable/disable supervisory console for M1250/2250 consoles with Enhanced Busy Lamp Field and Silent Observe.

5 LD 15 – Enable/disable an M1250/2250 console using Trunk Group Busy keys as status keys.

LD 15 – Enable/disable feature for an M1250/2250 console with a Console Graphics Module in the Standard Busy Lamp Field mode.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(XTG) ITG	Exclude/include Trunk Group Busy Indication.
- LFTN	 l s c u c u	Lamp Field Array Terminal Number For Large Systems For Small Systems and Succession 1000 systems
- SPVC	1-63 0	Attendant number for supervisory console. No supervisory console.
- - SBLF	(NO) YES	Supervisory lamp field array is not or is to be used to monitor other Attendant Consoles.
- ITH1	1-255	Visual indication threshold 1 (number of calls in queue Š ITH1 but < ITH2).
- ITH2	2-255	Visual indication threshold 2 (number of calls in queue Š ITH2 but < ITH3).
- ITH3	3-255	Visual indication threshold 3 (number of calls in queue Š ITH3).

LD 15 – Enable/disable feature for an M1250/2250 console with a Console Graphics Module in the Enhanced Busy Lamp Field mode.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(XBL) IBL	Exclude/include Busy Lamp Field or Console Graphics Module.
- - SPVC	1-63 0	Attendant number for supervisory console. No supervisory console.
- ITH1	1-255	Visual indication threshold 1 (number of calls in queue Š ITH1 but < ITH2).
- ITH2	2-255	Visual indication threshold 2 (number of calls in queue Š ITH2 but < ITH3).
- ITH3	3-255	Visual indication threshold 3 (number of calls in queue Š ITH3).

LD 12 – Enable/disable supervisory console Silent Observe.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	0 BVR	Add Busy Verify key (key 0) for silent observation.

LD 12 – Enable/disable supervisory console for M1250/2250 consoles with Enhanced Busy Lamp Field and Silent Observe.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
EBLF	(BLFD) BLFA	(Deny) allow Enhanced Busy Lamp Field.
KEY	0 BVR	Add Busy Verify key (key 0) for silent observation.

LD 15 – Enable/disable an M1250/2250 console using Trunk Group Busy keys as status keys.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(XTG) ITG	Exclude/include Trunk Group Busy Indication.
- SPVC	1-63 0	Attendant number for supervisory console. No supervisory console.
- - SBLF	NO	Supervisory lamp field array is not to be used to monitor other Attendant Consoles.
- ITH1	1-255	Visual indication threshold 1 (number of calls in queue Š ITH1 but < ITH2).

- ITH2	2-255	Visual indication threshold 2 (number of calls in queue Š ITH2 but < ITH3).
- ITH3	3-255	Visual indication threshold 3 (number of calls in queue Š ITH3).

Feature operation

Enable/disable Supervisory mode

To put your console in Supervisory mode, follow these steps:

1 Press  when your console is idle (all lpk indicators are off). Your console is now in Position Busy mode, preventing calls from ringing at your console.

2 To cancel Supervisory mode, press  again.

Monitor other attendants

In Supervisory mode, you can monitor selected attendant calls without being detected by either the attendant or the caller. To monitor an attendant, follow these steps:

1 Once in Position Busy mode, select an idle loop key.

2 Press **obs/B. ver.**

3 Dial the access code, then the attendant number:

a. If the called attendant is talking to a caller, you hear the conversation but you cannot be heard.

b. If the called console is idle, the S and D indicators go on.

c. If the called console is in Position Busy mode, you hear a fast busy tone, the S and D indicators flash quickly, and the OBS/B. VER indicator goes off.

4 Press **Rls** to end the procedure.

Call an attendant

To call an attendant in your group, follow these steps:

1 Once in Position Busy mode, select an idle **lpk** key.

2 Dial the attendant access code.

3 Dial the attendant code.
You hear ringing. The S indicator flashes slowly.

4 Press **Rls** to end the call.
The S indicator goes on steadily, and the RLS indicator goes on.

Transfer a call to an attendant

You can transfer a call to an attendant in your group, even if the attendant's console is in Position Busy mode. To transfer a call, follow these steps:

1 Dial the attendant access code; then the attendant code.
The EXCL SRC indicator goes on; the caller is automatically placed on hold. The D indicator flashes slowly, the lpk and S indicators are on.

a. If you dial an incorrect attendant code or if the called console is in Night Service mode, the transfer cannot be completed. You hear a fast busy tone and the D indicator remains off. Press **Rls**.

b. If the called console is busy, you hear a busy tone and the D indicator continues to flash slowly. Press **Rls** and your call is placed in the attendant queue.

2 Press the **lpk** key when the attendant answers.
The EXCL SRC indicator goes off and the D indicator lights steadily. You, the caller, and the attendant are connected.

3 Press **Rls** to end your connection in the call.

Assist an attendant

Even when your console is in Supervisory mode, an attendant can call you for assistance or transfer a call to you by following these steps:

1 You receive a call from an attendant while you are in Supervisory mode. You hear a tone. The S indicator flashes and the INTER POS. C. indicator goes on.

2 Press the **lpk** key next to the flashing S indicator.
The tone stops; the lpk and S indicators light steadily.
You are connected to the call.

Note: If it is a transferred call, the Call Waiting indicator lights. You must exit Position Busy mode to answer the call.

Attendant Trunk Group Busy Indication

Contents

This section contains information on the following topics:

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Feature interactions 386
Feature packaging 386
Feature implementation 387
Feature operation 387

Feature description

The attendant can control user access to a trunk route by pressing the appropriate Trunk Group Busy key. Station users with a Trunk Group Access Restriction (TGAR) from 0 to 7 accessing the route that has been busied out will be automatically intercepted to the attendant. Station users with a TGAR of 8 to 31 will not be affected and can dial out in the normal manner.

The Shift key allows the M1250 Attendant Console to have 16 Trunk Group Busy keys. The M2250 Attendant Console can have up to 20 Trunk Group Busy keys.

Trunk Group Busy Indication is allowed or denied on a customer basis. If allowed, the lamps associated with the Trunk Group Busy keys will provide visual indication of the status of the trunks within the route (See Table 17).

Table 17
Lamp states of Trunk Group Busy keys

Lamp state	Status of trunks
Off	Some of the trunks in the route are idle.
Flashing	All of the trunks in the route are busy.
Steadily lit	The attendant has taken control of the route.

Trunk Routes 0 to 9 are automatically assigned to keys 0 to 9 on the console.

On the M1250, Trunk Routes 0 to 15 are assigned 0 to 7 and 10 to 17 when the Shift key is activated. On the M2250, Trunk Routes are assigned to keys 0 to 9 and 10 to 19 when the Shift key is activated.

Operating parameters

There are no operating parameters associated with this feature

Feature interactions

Music

A music route that appears on a Trunk Group Busy key on the Attendant Console cannot be controlled by activation of the Trunk Group Busy key. In addition, the associated lamp will not reflect the status of the music trunks.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Allow Trunk Group Busy keys.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(IC1) IC2	Allow Trunk Group Busy keys, where: IC1 = 10. IC2 = 16 for M1250, or 20 for M2250.
- OPT	(XTG) ITG	(Exclude) include Trunk Group Busy Indicator keys.

Feature operation

To restrict access to a trunk route (make it busy to users):

- Press the **Trunk Group Busy** key associated with the trunk. The indicator goes on and remains steady.

To allow access to the trunk route:

- Press the **Trunk Group Busy** key associated with the trunk. The indicator goes off.

Audible Reminder of Held Calls

Contents

This section contains information on the following topics:

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Feature packaging 391
Feature implementation 391
Feature operation 392

Feature description

Occasionally, a user may forget that a call has been placed on hold. Audible Reminder of Held Calls (ARHC) allows an audible tone to operate as a reminder of a held call. It provides for a ring on analog (500/2500 type) telephones and a tone on proprietary telephones. The cadence and the duration between cadences are programmed per customer. This ability allows the user to differentiate between the cadence for Audible Reminder of Held Calls (ARHC) and the cadences of other existing features.

The station user will hear a ring or tone, which is repeated every 2 to 120 seconds depending on how this feature is programmed, as a reminder that a call is being held. A single-line telephone user must hang up after putting a call on Permanent Hold in order to start the timer.

Operating parameters

For analog (500/2500 type) telephones, Audible Reminder of Held Calls (ARHC) applies only to permanent hold. When using ARHC on a Meridian 1 proprietary telephone, the station user must not be originating, receiving, or active on another call.

Audible Reminder of Held Calls is supported on Multiple Appearance DNs; however, only the appearance initiating Hold will receive the reminder ring.

This feature does not operate on Attendant Consoles.

Feature interactions

Automatic Line Selection

The Audible Message Waiting signal is given if there is a message waiting on whatever line is selected by Outgoing Line Selection.

Call Hold, Permanent

Permanent Hold must be enabled in LD 10 for the single-line telephone; however, the ARHC timer takes precedence over the Permanent Hold timer.

On Hold on Loudspeaker

This feature works with the On Hold on Loudspeaker (OHOL) feature as for normal calls on hold (that is, gives a reminder there are calls on hold). Therefore, it is not recommended to use this feature with the OHOL feature.

Tones and Cadences

This feature allows for a definable cadence as a reminder of a held call. With an analog (500/2500 type) telephone, the cadence is determined by the customer's Flexible Tones and Cadence (FTC) table for the holding party. Ringing on an analog (500/2500 type) telephone is not affected by definitions for the Incoming Route option. The cadence for the reminder, and the duration between reminder rings, is always defined within the customer's tone table.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Set duration between reminder cadences for Audible Reminder of Held Calls.
2. LD 10 – Allow/deny Audible Reminder of Held Calls for analog (500/2500 type) telephones.
3. LD 11 – Allow/deny Audible Reminder of Held Calls for proprietary telephones.

LD 15 – Set duration between reminder cadences for Audible Reminder of Held Calls.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	TIM	Timers.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- DBRC	2-(60)-120	Duration between reminder cadences for Audible Reminder of Held Call. An odd numbered entry is rounded up to the next even number.

LD 10 – Allow/deny Audible Reminder of Held Calls for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	500/2500 telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA (ARHD) ARHA	(Deny) allow call transfer. (Deny) allow Audible Reminder of Held Calls.
FTR	PHD	Permanent Hold allowed.

LD 11 – Allow/deny Audible Reminder of Held Calls for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(ARHD) ARHA	(Deny) allow Audible Reminder of Held Calls.

Feature operation

No specific operating procedures are required to use this feature.

Authorization Code Security Enhancement

Contents

This section contains information on the following topics:

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Feature description

The Authorization Code Security Enhancement feature enables a user to temporarily override the access restrictions assigned to a station or trunk because of their assigned Network Class of Service (NCOS), Class of Service (COS), and Trunk Group Access Restrictions (TGAR) codes. If a user requires access to system facilities in addition to that allowed on the set, the Authcode feature can be used to provide them.

The Authorization Code (Authcode) Security Enhancement feature alerts the technician when an invalid Authcode is entered by generating an Authcode Alarm. The Alarm indicates to the technician that a valid user has inadvertently dialed the wrong digits or some unauthorized person may be trying to use an Authcode to illegally access the switch.

The Authcode Alarm is generated upon detection of violation of all Authcode-related features (that is, Basic, Network, and Station Specific Authorization code), except for calls originated by the attendant.

A new class of alarm has been added (Security Administration – SECA) to distinguish security violations from other types of system messages. The message SECA001 will be printed on the TTY indicating that an invalid Authorization Code has been dialed. The following is the format of the SECA001 message:

- Originated station or trunk Terminal Number
- Calling Line Identification (CLID) when the call is originated from an Integrated Services Digital Network (ISDN) trunk
- The Authorization Code entered

Operating parameters

This feature is enabled through the Authcode data block in LD 88.

The Authcode Alarm feature does not apply to calls originated by an attendant.

All existing operating parameters relating to Authorization Code usage apply to this feature.

All existing operating parameters relating to Fault Management apply to this feature.

For security reasons, the SECA001 alarm should not be configured in the Exception Filter table.

Feature interactions

Authorization Code Features

A Security Administration (SECA) message will be printed to the configured Maintenance Terminal (MTC), Filtered Alarm Output (FIL) console and/or the configured History File when an invalid Authcode is detected. The following features relate to Authorization Codes and are thus impacted: Basic Authorization Codes; Network Authorization Codes; Authcode Conditionally Last; Direct Inward System Access with Authorization Code; Station Specific Authcode; Speed Call/Autodial with Authorization Codes; Call Forward with Authorization Codes; Scheduled Access Restrictions with Authorization Codes; Network Queuing/Remote Virtual Queuing with Authorization Codes; Coordinated Dialing Plan with Authorization Codes; and Flexible Feature Code with Authorization Codes.

Charge Account, Forced

If the Authorization Code is used to change the Class of Service of the user, the new Class of Service must be TLD, CTD, or CUN. If an Authorization Code entered after FCA has altered the Class of Service to unrestricted (UNR), the change made by the Authorization Code still comes into effect.

If the originator's Network Class of Service (NCOS) has been changed by an Authorization Code prior to an applicable FCA entry, the new NCOS is replaced by the FCA NCOS, provided the new Facility Restriction Level (FRL) is not lower than the existing FRL. Similarly, if the originator's NCOS has been changed by an FCA entry, the NCOS will be changed again by a valid Authorization Code entry.

China - Flexible Feature Codes - Outgoing Call Barring

Digits dialed after an Authorization Code are checked against the active Outgoing Call Barring level.

Direct Private Network Access with Authorization Code Retry

Only when an Authcode retry fails will a SECA message be printed to the configured MTC, FIL console and/or the configured History File.

Last Number Redial

These codes are not stored in Last Number Redial (LNR). To use these features when calling the number stored in LNR, the code must first be dialed manually. When dial tone is returned, LNR can be used to complete the dialing.

New Flexible Code Restriction

If the Class of Service of the authorization code is Toll Denied (TLD), NFCR is applied. If the Class of Service is Conditionally Unrestricted (CUN) or Conditionally Toll Denied (CTD) and the call is not routed through BARS/ NARS, CDP or ANI, NFCR is applied.

Pretranslation

The first digit dialed after a valid Authorization Code is sent to the pretranslator.

Scheduled Access Restrictions

Authorization Codes can be used to override Scheduled Access Restrictions. In addition, Authorization Codes are defined for the specific use of SAR FFCs.

Speed Call, System

If the Basic Authorization Code (BAUT) or Network Authorization Code (NAUT) package is equipped, a Network Class of Service (NCOS) is assigned to the System Speed Call list. The NCOS of the System Speed Call list replaces the NCOS of the Authorization code or Forced Charge Account code if it increases the Facility Restriction Level (FRL) of the code.

Station Specific Authorization Code

Users cannot freely enter authorization codes from telephones that have AUTR or AUTD Class of Service.

Stored Number Redial

The Authorization code is not stored. To store a code, dial the code prior to using Stored Number Redial to dial the call.

Feature packaging

This feature is included in base System Software.

The following software packages are optional, but may be needed depending upon the application:

- Alarm Filtering (ALRM_FILTER) package 243
- Basic Authorization Code (BAUT) package 25
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Direct Private Network Access (DPNA) package 250
- Direct Inward System Access (DISA) package 22
- Network Class of Service (NCOS) package 32
- Network Authorization Code (NAUT) package 63
- Station Specific Authcodes (SSAU) package 229
- Recorded Announcement (RAN) package 7
- Scheduled Access Restrictions (SAR) package 162
- System Speed Call (SSC) package 34, or Network Speed Call (NSC) package 39

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 88 – Configure the Authcode Alarm for each customer.

2 LD 17 – Configure the Alarm Filter.

LD 88 – Configure the Authcode Alarm for each customer.

Prompt	Response	Description
REQ	NEW CHG	Configure or change.
TYPE	AUB	Authcode Data Block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password.
ALEN	1-14	Number of digits in Authcode.
ACDR	(NO) YES	(Do not) activate CDR for authcodes.
AUTHCOD_ALRM	(OFF) ON	(Disable) enable Authcode Alarm.
RANR	0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems

LD 17 – Configure the Alarm Filter.

Prompt	Response	Description
REQ	CHG	Change configuration.
TYPE	ALARM	Alarm Filters.
- FMT_OUTPUT	ON	Enables formatting for the alarm/exception output. <CR> retains current formatting status.
- AF_STATUS	ON	Alarm and Exception filtering.
- SUPPRESS	0-(5)-127	Alarm occurrence threshold (prior to suppressing). Determines number of times an alarm may occur before it is no longer output. The entry 0 indicates that all alarm occurrences are output (no suppression).

- ESCALATE	0-(2)-127	Alarm occurrence threshold (prior to escalating). Determine number of occurrences of alarm before it is escalated to critical severity over a 24 hour period when the Interval Time has elapsed, the Interval Alarm counter is cleared. Applicable only to Major alarms. The entry 0 occurrence disables alarm escalation.
- A_FILTER	NEW CHG OUT	Add, Change, or Remove an Alarm Filter entry.
- TRIGGER	SECA001	Alarm report consisting of the mnemonic “SECA” and the numerics “001” must be entered for the Authcode security alarm.
- SEVERITY	aaaa	Alarm severity of a particular alarm entry, where: NONE = no rating and default status CRITICAL = System operation is in jeopardy MAJOR = Serious condition, system is operational MINOR = Error condition detected, system operation not affected <CR> = current value retained

Feature operation

No specific operating procedures are required to use this feature.

Autodial

Contents

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Feature description

Autodial (ADL) allows users to dial a number by pressing a single key. Proprietary telephones and Attendant Consoles can be assigned an Autodial key/lamp pair.

The number stored against the Autodial key can be programmed or changed at any time. The maximum number of digits the user is allowed to program can be 4, 8, 12, 16, 20, or 23 digits. Depending on the length allowed, the Autodial number can be another DN or an access code plus further digits. The asterisk (*) can be used as a pause for outpulsing (that is, for outgoing trunks) when required. When the Autodial key is pressed, the stored number is processed as if it had been dialed manually.

Speed Call/Autodial with Authorization Code. This enhancement allows an Authorization Code to be included in a Speed Call entry or an Autodial key. Entries can contain any one of the following combinations:

- SPRE code + digit 6 + authorization code
- SPRE code + digit 6 + authorization code + #, or
- SPRE code + digit 6 + authorization code + # + Electronic Switched Network (ESN) access code and dialed number.

Autodial Flexible Feature Codes
A user can define an Autodial DN that is automatically dialed by the system in one of two ways:

- In LD 10, while defining the Autodial DN length under the feature (FTR) ADL.
- Using the Autodial Activate (ATDA) FFC, defined in LD 57. This method requires that the length of the Autodial must first be defined in LD 10. The user goes off hook and dials the ATDA FFC. Upon receiving dial tone, the user enters the desired Autodial DN, and then goes on hook.

If, after going off hook, no digits are entered within a customer-defined period of time (defined in LD 15) under ADLD (Autodial Delay), the Autodial DN is automatically dialed.

> ***Note:*** In LD 10, the user can define a partial DN as an Autodial DN. The user can enter the remaining digits while making a call – the user goes off hook, waits for the dial tone to time out, and then enters the remaining digits of the desired DN. The call is then dialed out.

To deactivate Autodial, the user dials either the Autodial Deactivate (ATDD) FFC (defined in LD 57) or the general Deactivate (DEAF) FFC (also defined in LD 57).

Operating parameters

Autodial must be assigned to a key/lamp pair. As a result, it is not available on analog (500/2500 type) telephones.

To use Autodial, the Autodial Activate (ATDA) FFC must have been entered previously and an Autodial number must be stored.

An attendant can enter an Authorization Code for other callers provided that the system is equipped with the Network Authorization Code (NAUT) package.

On Attendant Consoles, pressing the Autodial key, then pressing a Speed Call key is not allowed.

Authorization Code Conditionally Last is not supported by the Autodial feature.

An octothorpe (#) is required as a delimiter after the Authorization Code if an ESN access code and dialed number is stored as part of the Autodial key. If the octothorpe is not entered, the user receives fast busy tone. The octothorpe is not stored in the CDR record.

The Autodial feature allows a maximum of 23 digits including the SPRE code, the digit 6, the Authorization Code, the delimiter (#), the ESN access code, and the dialed number.

If the system initializes before the Authorization Code is recorded by CDR, the record will be lost.

An SL-1 digit display telephone can display up to 16 digits. Additional digits cause the digits to scroll off the display.

Because it has a Directory, the M3000 Touchphone does not support the Autodial feature.

On digit display telephones, Authorization Codes cannot be blocked from being displayed.

The Authorization Code is not validated during the storing process. An invalid Authorization Code is detected when the Autodial key is activated.

Network Automatic Route Selection (NARS) and Basic Alternate Route Selection (BARS) does not support the asterisk (*) as a pause when dialing an autodial number.

Feature interactions

AC15 Recall: Transfer from Meridian 1

Autodial and Last Number Redial are supported with the AC15 Recall: Transfer from Meridian 1 on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone detection is performed by the system.

Additional transfers are supported if the stored digits are outpulsed without any treatment. For example, a route is seized and the route access code is outpulsed to the far end and interpreted as a Directory Number. No dial tone detector or timer is started, so the digits are outpulsed immediately without checking the state at the far end.

Automatic Redial

Automatic Redial can be activated on a dialed number using the Autodial (ADL) key.

Call Forward and Busy Status

Party A can use the Busy/Forward Status (BFS) key as an Autodial key to dial party B.

Call Party Name Display

No name information displays during the programming of Autodial numbers.

Calling Party Privacy

An outgoing trunk call initiated by pressing the Autodial key will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored against the Autodial key. The CPP code is counted against the maximum number of digits (currently 23) stored against the Autodial key.

A user can also store the CPP code against the Autodial key. An outgoing CPP call can be initiated by pressing the Autodial key, followed by manually dialing the digits.

An outgoing CPP call can also be initiated by dialing the CPP code, followed by pressing the Autodial key against which the normal dialing sequence of digits have been stored.

Charge Account and Calling Party Number

Charge account numbers, including the Charge Account access Special Prefix (SPRE) code, can be stored as Speed Call or Autodial numbers. All current limitations of these features apply, such as a maximum of 23 digits per entry, including the access code. An Autodial number or dialed digits can follow, but not precede, a Speed Call number. The digits generated by an Autodial key during feature operation are accepted as Charge Account digits.

Charge Account, Forced

Forced Charge Account (FCA) numbers (including the Special Prefix [SPRE] code and the Charge Account access code) can be entered in Speed Call lists or stored as Autodial numbers. The digits can also be stored, provided that the account number, regardless of its length, is followed directly by an octothorpe (#).

China – Flexible Feature Codes - Busy Number Redial Enhanced Flexible Feature Codes - Busy Number Redial

Activation of Busy Number Redial (BNR) changes the activation of Autodial. The DN that is autodialed becomes the DN that was busy. When the BNR activation timer expires or the busy DN is redialed when it is idle, the autodial capability is deactivated, but the number saved is not cleared. If Autodial is then activated without entering a DN, the number used is the formerly busy DN.

Activation of Autodial when BNR is active deactivates BNR.

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Autodial.

Dial Intercom

The Dial Intercom code can be dialed using Autodial or Speed Call.

Direct Private Network Access

If Autodial is programmed with a valid Authcode for Authcode Last component of Direct Private Network Access followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be prompted for the Authcode again.

Flexible Hot Line
Enhanced Hot Line

Flexible Hot Line and/or Enhanced Hot Line are mutually exclusive with the Autodial feature.

Intercept Computer Dial from Directory

It is possible to press the Autodial (ADL) key (in which some digits are stored such as an Electronic Switched Network (ESN) code or Flexible Feature Code (FCC)), and then dial a DN from the Intercept Computer. The DN will then be stored on the ADL key.

Last Number Redial

A number dialed using Autodial will become the Last Number Redial number on all telephones, except the M2317 and M3000.

Station Specific Authorization Code

The Station Specific Authorization Code (SSAU) feature treats stored autodial numbers as if they were entered at the telephone.

Speed Call Delimiter

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Autodial on a E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange/Central Office, which is expected to appear after the toll access code is dialed. The Public Exchange then does not accept the call due to failure to receive ANI information.

User Selectable Call Redirection

User Selectable Redirection Allowed (USCR) does not support Autodial; Autodial cannot be used to dial all or part of the digits for USCR programming.

Feature packaging

Optional Features (OPTF) package 1 includes Autodial and has no feature package dependencies.

To implement Autodial with Authorization Code, the following packages are required:

- Charge Account/Authorization Code Base (CAB) package 24, or
  Basic Authorization Code (BAUT) package 25, or
  Network Authorization Code (NAUT) package 63.
- Optional Features (OPTF) package 1, or
  System Speed Call (SSC) package 34, or
  Network Speed Call (NSC) package 39.

The following packages are required for Autodial FFCs:

- Flexible Feature Codes (FFC) package number 139, and
- Background Terminal Facility (BGD) package 99.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Assign Autodial key for proprietary telephones.

2 LD 12 – Assign Autodial key for M1250 and M2250 Attendant Consoles.

3 LD 15 – Define Autodial Delay in the Customer Data Block.

LD 11 – Assign Autodial key for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx ADL yy zzz...z	xx = assigned key number. yy = the length of the Autodial number (4, 8, 12, 16, 20, or 23 digits; default is 16). zzz...z = the digits to be dialed automatically (optional).

LD 12 – Assign Autodial key for M1250 and M2250 Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	1250 2250	Console type.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx ADL zzz...z	xx = assigned key number. zzz...z = the digits to be dialed automatically (optional).

LD 15 – Define Autodial Delay in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FFC	Flexible Feature Codes
CUST	0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- ADLD	(0)-20	Autodial Delay, in seconds. If 0, then FFC Autodial for 500/2500 telephones is disabled. Only prompted if FFC package (139) is equipped. Inputs are rounded up to the next valid increment of two (that is, input of 11 would be rounded up to 12).

Feature operation

To program Autodial, follow these steps:

1. While the handset is on hook, press the **Autodial** key.
   The associated lamp flashes.
2. Dial the desired number and press the **Autodial** key again.
   The lamp goes dark.

To use Autodial, follow these steps:

1. Lift the handset off hook, or press the **Handsfree** key if allowed.
2. Press the **Autodial** key.
   The call is dialed.

The following instructions are for using the Autodial FFCs:

- Activate and program
  The user must dial the Autodial Activate (ATDA) FFC followed by the number to be stored as the Autodial number.
- Activate only
  The user must dial the Autodial Activate (ATDA) FFC.
- Deactivate
  The user must dial the Autodial Deactivate (ATDD) FFC or the Deactivate (DEAF) FFC.
- Use
  The user goes off hook, if no digits are dialed within the customer defined time period (ADLD), the system then dials the number stored as the Autodial number.

  Note: To use Autodial, the Autodial Activate (ATDA) FFC must have been entered previously and an Autodial number must be stored.

Autodial Tandem Transfer

Contents

This section contains information on the following topics:

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Feature description

Prior to the introduction of this feature, in order to access the Central Office (CO) transfer feature after a Centrex/Trunk Hook Flash on an established trunk call, the user had to manually dial the digits. This procedure permits call completion, but is slow and requires knowledge of the full telephone number. The Autodial Tandem Transfer (ATX) feature allows the Autodial key to be used after a switchhook flash to out pulse Dual-tone Multifrequency (DTMF) digits while a call is in an established state.

One application for the Autodial Tandem Transfer feature is for use in a 911 environment to transfer an emergency call from a Public Safety Answering Point (PSAP) to the most appropriate participating emergency agency. Manually dialing the digits by the PSAP in order to transfer the 911 call to another PSAP can take time and is subject to misdialing. To avoid this, the ADL key programmed with the special station number can be used to send digits to the tandem/Centrex office to transfer the call. Using the ATX feature, a PSAP can transfer the incoming call by pressing the Trunk Hook Flash (THF) key, waiting for a broken dial tone, and then pressing the ADL key.

Operating parameters

The Centrex/Trunk Switchhook feature only supports voice calls. Subsequently, the ATX feature which uses Centrex/Trunk Hook Flash does not support data calls.

Centrex/Trunk Hook Flash cannot be activated during Conference and No Hold Conference calls. Subsequently, the ATX feature which uses Centrex/ Trunk Hook Flash does not support them either. Only two-party calls are supported by the ATX feature.

The following trunk types are supported by the ATX feature: AID, CAA, CAM, COT, TIE (supports ATX, not Trunk Hook Flash), CSA, DID, DOD, WATS, DTI, and DTI2.

The ATX feature is not supported on analog (500/2500 type) telephones, Attendant Consoles, and BRI sets.

End-to-End signaling (EES) is not supported for this feature (only Improved End-to-End signaling is supported).

Single CPU machines are not recommended for 911 applications. Meridian 911 hardware may be required for 911 applications.

Feature interactions

Automatic Dial

The ADL key is used by the Automatic Dial feature to send DN digits out during the dialing stage. Some of the digits, such as “#” and “*”, have special meanings. The “*” causes a three-second pause, while the “#” means end of dialing.

In the ATX feature when the ADL key is used during an established call, the DTMF tones corresponding to the digits programmed in the ADL key are sent out (using End-to-end Signaling to send the digit out). Therefore, the DTMF tones corresponding to “#” and “*” are outpulsed.

Call Detail Recording

No modifications to this feature are required for the ATX feature.

For 911 applications, most of the calls are incoming calls. The outgoing End-to-End Signaling digits are captured for incoming 911 calls on the incoming CDR records. This only applies to 911 trunks.

Centrex Switchhook Flash

Because Autodial Tandem Transfer uses Centrex Switchhook Flash (THF), it is affected by any modification to the THF enhancement feature.

Conference

The ATX feature is blocked during Conference and No Hold Conference calls.

Digit Display

Digit Display allows the automatic display of information relevant to normal call processing if the sets have display capability and the Class of Service is ADD or DDS. When the THF key is pressed, the display gets cleared, and pressing the ADL key causes the ADL digits to be displayed. However, no ADL digits will be displayed if no Tone and Digit Switch (TDS)/XCT is available to generate the Dual-tone Multifrequency (DTMF) tones for the ADL digits.

End-to-End Signaling

EES is used to send the Automatic Dialing (ADL) digits to the Public Exchange/Central Office (CO). With Autodial Tandem Transfer (ATX), the 911 agent can use the ADL key or manually dial the digits, or use a combination of both methods, to dial the third party's number. The ADL key can be pre-programmed with a prefix and the remaining digits can be dialed manually to distinguish between different numbers. When you combine manual dialing with the ADL key, if EEST = YES and DTMF = YES in LD 15, you hear the DTMF feedback tone as a result of manual dialing and a single feedback tone as a result of pressing the ADL key. To get uniform feedback tone when using the ADL key along with manual dialing, set the DTMF prompt to NO in LD 15.

Improved End-to-End Signaling is used to send the pre-programmed ADL digits to the CO. With the ATX feature, a 911 Agent can use the ADL key, or manually dialed digits, or a combination of both to dial the third party's number. It is recommended to set the DTMF prompt to NO (EES – LD 15) to get uniform feed back tone (single feed back tone) when using the ADL key along with manual dialing.

Last Number Redial

Normally, when the ADL key is pressed during the dialing stage, the ADL number will replace the Last Number Redial number. In the ATX feature, however, when the ADL key is used during the established stage, the ADL digits will not substitute the Last Number Redial number.

Malicious Call Trace - Enhanced

Enhanced Malicious Call Trace implements the ability to send a call trace request to the CO and provides the possibility to record the call using a recorder. This feature also uses the Centrex/Trunk Switchhook Flash feature; the same enhancement applies to the ATX feature.

Speed Call

The Speed Call key cannot be used after THF or during an established call to send digits out to the far site; it can only be used during the dialing stage.

Feature packaging

Autodial Tandem Transfer (ATX) is package 258.

The following packages are also required:

- End-to-End Signaling (EES) package 10
- Trunk Hook Flash (THF) package 157

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Define THF and ADL keys for Meridian 1 proprietary telephones.

2 LD 14 – Define THF Class of Service THFA for the trunk.

3 LD 15 – Define feedback tone when ADL digits are sent out.

4 LD 16 – Set the duration for Centrex/Trunk Switchhook Flash.

LD 11 – Define THF and ADL keys for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type where xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

CLS	...	
KEY	xx THF	Key xx is configured for the Centrex/Trunk Switchhook Flash feature.
	yy ADL ll zz..zz	Key yy is configured for the Autodial key; ll is the length of the autodial number (the default is 16). zz..zz are the digits to be dialed automatically.

LD 14 – Define THF Class of Service THFA for the trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaa	Trunk type, where: aaa = AID, CAA, CAM, COT, TIE (supports ATX, not Trunk Hook Flash), CSA, DID, DOD, WATS, DTI, and DTI2.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(THFD), THFA	The THF feature is (denied) allowed; the default is THFD.

LD 15 – Define feedback tone when ADL digits are sent out.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	CDR	Call Detail Recording.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

CDR ...	YES	Call Detail Recording.

- ECDR	YES	Include EES digits in CDR record. This will include ADL digits that are outpulsed during an established call.
...		All-digital connection prefix.
TYPE:	FTR	Features and Options.
EEST	YES NO	End-to-end Signaling feedback tone to originating party.
- DTMF	YES NO	Single feedback tone is provided for the user. ***Note:*** With a Yes or No response, single tone feedback is only available.

LD 16 – Set the duration for Centrex/Trunk Switchhook Flash.

Prompt	**Response**	**Description**
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKYP	aaa	Trunk type.
...		

CNTL	(NO) YES	Change control or timers.
- TIMR	FLH <space> 60-(510)-1536	Flash timer in msec. The range of the Centrex switchhook flash timer is 60-(510)-1536. The FLH value is rounded down to the nearest 10 msec. tick. If the value entered is 128 or 129, then it is set to 130 msec. *Software controlled flash* 60-127 msec. Digit 1 will be sent. 128-1536 msec. software controlled switchhook flash. ***Note:*** An FLH timer value of 127 msec. or less is not supported by the XFCOT card. The firmware controlled flash must be used. *Firmware controlled flash* The user can enter any value from 60 to 1536 msec. 90 msec. is the hardcoded firmware flash for an XFCOT card; the technician should enter 90 msec. ***Note:*** The FWTM prompt must be set to YES for the trunk associated with this route in LD 14, if firmware timing is to be used.

Feature operation

Normal operation

1 An incoming call from a Central Office (CO) terminates to a proprietary telephone.

2 The set user presses the **THF** key, waits for a broken dial tone from the CO, and then presses the **ADL** key to send a string of digits to the CO (the ADL has been pre-programed with the number).

3 The CO will transfer the call to the third party dialed by set A.

Meridian 911 operation

1 An incoming 911 trunk call to a tandem/Centrex office terminates to a PSAP on the system.

2 The PSAP call taker presses the **THF** key, waits for a broken dial tone, and then presses the **ADL** key to call the proper number (such as a police station).

3 The PSAP call taker then disconnects to complete the transfer.

Automatic Answerback

Contents

This section contains information on the following topics:

Feature description 419
Operating parameters 419
Feature interactions 420
Feature packaging 421
Feature implementation 421
Feature operation 422

Feature description

Automatic Answerback (AAB), when assigned to a Meridian 1 proprietary telephone, allows any incoming call to a single appearance Prime Directory Number (PDN) to be answered automatically. An incoming call will ring one time, then the system will turn on Handsfree and establish a speech path. When either party hangs up, the call is automatically disconnected.

Automatic Answerback can be permanently assigned either as a Class of Service, or with an Automatic Answerback key/lamp pair assigned to allow activation/deactivation of the feature. If privacy is desired during a call, handset operation is allowed.

Operating parameters

This feature is available on M2112, M2317, and M2616 telephones.

Incoming ground start trunks must provide Answer Supervision. If not, the call is connected to the attendant who provides the necessary supervision.

The Prime DN (PDN) must be a single appearance DN.

Calls presented to DNs other than the PDN, or calls presented to the PDN when active on another DN, will not receive Automatic Answerback treatment.

Automatic Answerback can be provided as a Class of Service or on a key/lamp pair. You cannot assign both in service change.

Feature interactions

Automatic Line Selection

Automatic Answerback operates only on the Prime DN (key zero) and has no interrelation with Incoming Ringing/Non-Ringing Line Selection.

Called Party Disconnect Control

Incoming calls on a trunk with Called Party Disconnect Control Allowed that terminate on a telephone with Handsfree Answerback are answered automatically. They are not disconnected automatically, however, when the calling party goes on-hook.

Collect Call Blocking

The Automatic Answerback (AAB) feature, when assigned to a Meridian 1 proprietary telephone, allows any incoming call to a single-appearance Prime Directory Number (PDN) to be answered automatically. If an incoming DID or CO call terminates on a set with the AAB feature enabled, the call is automatically answered after one ring. If the set has a CCBA Class of Service, the CCB answer signal is provided in the place of the regular answer signal.

Hot Line

The Automatic Answerback feature is fully compatible with a two-way Hot Line key assigned as the Prime DN.

Message Center

If a telephone is in the Automatic Answerback mode, incoming calls are not routed to the Message Center.

Feature packaging

Automatic Answerback (AAB) package 47 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Assign Automatic Answerback as a Class of Service to SL-1, M2112, M2317, M2616, or M3000 telephones.

2 LD 11 – Assign Automatic Answerback key to SL-1, M2112, M2317, M2616, or M3000 telephones.

LD 11 – Assign Automatic Answerback as a Class of Service to SL-1, M2112, M2317, M2616, or M3000 telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2112, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(AAD) AAA (HFD) HFA	(Deny) allow Automatic Answerback for all calls. AAA cannot be entered if the AAK key is already programmed. (Deny) Allow Handsfree. ***Note:*** HFA is allowed for the M2216 only.

LD 11 – Assign Automatic Answerback key to SL-1, M2112, M2317, M2616, or M3000 telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2112, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(HFD) HFA	(Deny) allow Handsfree. ***Note:*** HFA is allowed for the M2216 only.
	(AAD) AAA	Allow (Deny) Automatic Answerback. Must disable to add the AAK key.
KEY	xx AAK	Add Automatic Answerback key. xx = key number. ***Note:*** The M2216 with AAA cannot use key 5 as a feature key. Key 5 is reserved for handsfree. The M3000 must use key 35.

Feature operation

To activate Automatic Answerback, follow this step:

- Press **Auto Answer**.
  Incoming calls to your PDN will ring once, then be answered with Handsfree turned on.

To deactivate Automatic Answerback, follow this step:

- Press **Auto Answer**.
  Incoming calls to your PDN will not be answered automatically.

  Note: If Automatic Answerback is assigned as a Class of Service instead of a key on your telephone, you cannot deactivate it.

Automatic Call Distribution

Automatic Call Distribution (ACD) is an optional feature. The ACD feature is used when a large number of incoming calls are answered by a group of ACD-assigned telephones. Incoming calls are served on a first-in, first-out basis and are distributed among the available telephones so that the agent position that has been idle the longest is provided with the first call. This guarantees that incoming calls are distributed equally to all agents.

Consult the following NTPs for information regarding the ACD feature:

- *Automatic Call Distribution: Description* (553-3001-351)

Automatic Gain Control Inhibit

Contents

This section contains information on the following topics:

Feature description 425
Operating parameters 425
Feature interactions 426
Feature packaging 426
Feature implementation 426
Feature operation 426

Feature description

The Automatic Gain Control (AGC) function, supported by the A44 chip in Meridian digital sets, lowers handset sound levels to minimize background noise. The AGC Inhibit enhancement allows a customer to suppress this function, on a system basis.

Whenever a transmission download occurs, which happens following a SYSLOAD or when the set line cord is plugged in, the option setting in LD 17 is included in the message. The message is interpreted by set's firmware and the appropriate setting is applied.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 17 – Define the AGC setting.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATRN	Aries Transmission.
...		
ATRN	YES	Aries (Meridian Modular set) transmission parameter; only prompted if the response to TYPE is CFN.
...		
- AGCD	(NO) YES	Automatic Gain Control Disable.

Feature operation

No specific operating procedures are required to use this feature.

Automatic Guard Detection

Contents

This section contains information on the following topics:

Feature description 427
Operating parameters 427
Feature interactions 428
Feature packaging 428
Feature implementation 428
Feature operation 428

Feature description

This feature verifies the transition from a high-resistance to a low-resistance loop upon correct seizure of an inactive trunk. Incorrect seizure results in the release of the faulty trunk and the attempted seizure of the next trunk in the hunt sequence.

Automatic Guard Detection will prevent the seizure of a trunk if the trunk:

- is an open circuit in tip, ring, or both; or
- has no current present when the trunk is seized

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

LD 14 –Enable or Disable Automatic Guard Detection for outgoing trunks.

Prompt	Response	Description
REQ	New CHG	Add, or change
TYPE	a...a	Type of truck.
SEIZ	(NO) YES	Automatic Guard Detection for outgoing trunks (disabled) enabled.

Feature operation

No specific operating procedures are required to use this feature.

Automatic Hold

Contents

This section contains information on the following topics:

Feature description 429
Operating parameters 430
Feature interactions 430
Feature packaging 433
Feature implementation 433
Feature operation 434

Feature description

The Automatic Hold feature allows an active call to be put on hold without having to use a separate Hold key. There are three ways to put a call on hold with the Automatic Hold feature:

- Press the active call key. The established call is automatically placed on hold.
- Press an idle Directory Number (DN) key. The established call is automatically placed on hold.
- Press any idle key and the established call is placed on hold.

If a set user is on an established call and wishes to answer an incoming call or initiate an outgoing call, the set user can press any idle DN key to place the call on hold and either initiate or establish a call on the same key. To terminate a call with the Automatic Hold feature, the Release key must be pressed.

This feature requires a new Class of Service implementation (Automatic Hold Class of Service).

Operating parameters

The Automatic Hold feature can be equipped on all multi-line proprietary sets. The functionality to hold a call already exists on the Attendant Console. The Automatic Hold feature is not applicable on analog (500/2500) type sets.

Feature interactions

Attendant Break-In to Inquiry Calls

A consultation call on proprietary sets, using a second DN along with Automatic Hold, is not treated as an inquiry call. The consultation call may be broken-in to, but the call held on the first DN is not involved in the Break-In.

Attendant Recall with Splitting

Automatic Hold does not have precedence over Attendant Recall (for instance, Automatic Hold cannot be activated until the attendant answers the recall presented on the console). However, it can be activated before the attendant answers a call transferred to the console.

Automatic Call Distribution Incalls Key

Automatic Call Distribution (ACD) does not override the Incall 5 key. The Incalls key is unique to the Automatic Hold feature. If an Automatic Call Distribution (ACD) agent has an active call on the Independent Directory Number (IDN) key, and a call comes in to an Incalls key, pressing the Incalls key to answer the call puts the active call on the IDN key on hold.

Call Transfer

If a call is established or ringing on the Transfer key, pressing any idle DN key automatically puts the call on hold. To transfer an active call, press the transfer key once to reestablish the call, press a second time to complete the transfer. To release the transfer feature you must press the release key.

Call Waiting

Pressing the Call Waiting key to answer a waiting call, makes that call active while the previous call is put on hold.

Conference

If a call is established on the conference key, pressing any DN key puts the Conference call on hold. The user must press the conference key to reestablish the call. Pressing the conference key a second time completes the Conference call.

No Hold Conference

The Automatic Hold feature does not apply in the case of a No Hold Conference call. Automatic Hold does not override the No Hold Conference feature.

Digit display

Digit display is the same with automatic hold as it was with manual hold.

Automatic Answer Back

The Automatic Hold feature is not applicable with the Automatic Answer Back feature.

Individual Hold Enhancement

When a Multiple Appearance Directory Number (MADN) call is put on hold on proprietary sets, the Hold key lamp flashes at this user's set, while a slow flicker is shown at all other appearances of the same DN. With more than one single line MADN (SCR/SCN/HOT/PVR/PVN) active on a conference call, the user is put on hold either by pressing the Hold key, or with Automatic Hold feature enabled, the user can press the active single line MADN. With the Release option disabled, the active call on the single line MADN is put on hold. With the Release option enabled, the active call on the single line MADN is dropped.

Display Overflow on Calling Number Identification

If the number of Calling Number Identification (CNI) digits exceeds the capacity of the digit display, the active DN key can be pressed to show the remaining digits. If the active DN key is pressed again, the established call is placed on hold. The established call can be placed on hold, before the digits are displayed, by pressing any other DN key.

Group Call (GRC)

Only the originator of a Group Call (GRC) can put the Group Call on hold.

Hold Key

A set configuration with Automatic Hold Allowed Class of Service can still place calls on hold using the Hold key.

Enhanced Hotline and Hotline No Hold Conference

On proprietary sets pressing a designated Hotline key places an outgoing call to a pre-defined DN. Pressing any idle DN key or pressing the hotline key a second time can place this call on hold. The user can use the same DN key they used to put the call on hold to make an outgoing call or to answer an incoming call.

On a two-way Hotline key, the incoming call is held if the hotline key is pressed twice or if an idle DN key is pressed. Pressing the Release key while on an active Hotline call terminates the call.

The Conference-Hot Line (CH) key does not support Automatic Hold.

Lamp Status

The LED lamp status indications of calls put on automatic hold are identical to those for calls that are put on hold using the Hold key.

Last Number Redial (LNR)

A set with Last Number Redial Allowed (LNA) Class of Service can put an active call on hold by pressing another idle DN key and still activate the Last Number Redial feature to make an outgoing call. Automatic Hold does not override this feature.

Music on Hold

Music on Hold can be applied to calls put on automatic hold.

Voice Call

If a user presses the Voice Call key while a call is established on the key, the call is placed on hold. If the Voice Call key is pressed while a call is established on another DN, the established call is put on hold.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Allow or deny the Automatic Hold Class of Service for proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Set type xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15.
...	...	
CLS	AHA	AHA = Automatic Hold allowed. (AHD) = Automatic Hold (denied).

...	...	
KEY	xx aaa yyyy	The set type must be configured with two DN keys. Where: xx = key number aaa = DN key type. DN types supported include: ACD, CWT, DIG, GRC, HOT, MCN, MCR, SCR, SCN, or VCC. yyyy = Directory Number for key type. ***Note:*** Refer to feature interactions in this chapter when assigning keys to see if feature operation conditions are affected.

Feature operation

Put a call on hold

With Automatic Hold enabled, a call can be placed on hold by pressing the DN on which the call is active or by pressing any other idle DN key.

Make a new call

An active call can automatically be placed on hold, if any idle DN key is pressed. A new call can now be made on the DN key that was pressed or any other DN key.

Answer a call

If the set user is on an active call and a second call is presented on another DN, the user can answer the incoming call which automatically places the first call on hold.

A user of a set having Automatic Hold Class of Service can still place an active call on hold by pressing the Hold key.

Terminate a call

To terminate a call the set user must press the Release key.

Automatic Line Selection

Contents

This section contains information on the following topics:

Feature description 435
Operating parameters 436
Feature interactions 436
Feature packaging 438
Feature implementation 438
Feature operation 438

Feature description

Automatic Line Selection allows manual or automatic selection of incoming and outgoing lines for a given Meridian 1 proprietary telephone on a Class of Service basis. When a user lifts the handset, the telephone automatically selects a preferred line according to its priority. The line preferences are as follows, listed in order of selection priority:

- Manual Line Selection
  The user manually selects the DN to be used before going off-hook. Dial tone is returned if the line is idle. If the line is ringing, the call is answered and connected to the speaker of the telephone or Handsfree unit.

- Incoming Ringing Line Selection
  With Incoming Ringing Line Selection enabled, when the user goes off-hook, the telephone automatically scans the DN keys (without the user first manually selecting a DN key). If a line on the telephone is ringing, it is selected and the call is answered.
- Incoming Non-Ringing Line Selection
  With Incoming Non-Ringing Line Selection enabled, when the user goes off-hook, the telephone scans the DN lines and answers any unanswered incoming calls that appear but do not ring at that telephone.
- Outgoing Line Selection
  With Outgoing Line Selection enabled, when the user goes off-hook, the telephone scans the DN keys for an idle line. If a line is idle, it is selected and dial tone is returned.
- Prime Line Selection
  When the handset is lifted, the system processes any manual, incoming, or outgoing line selections. If no line is selected by one of these modes, a designated Prime Line (the DN on key 0) is selected.

Operating parameters

The Automatic Line Selection feature is available on Meridian 1 proprietary telephones only.

The user determines which line is in use by observing lamp state changes.

Feature interactions

Audible Message Waiting

The Audible Message Waiting signal is given if there is a message waiting on whatever line is selected by Outgoing Line Selection.

Automatic Call Distribution (ACD)

An ACD DN is not selected by automatic Incoming Non-Ringing and Outgoing Line Selection. It is selected by Incoming Ringing Line Selection.

Automatic Answerback

Automatic Answerback operates only on the Prime DN (key zero) and has no interrelation with Incoming Ringing/Non-Ringing Line Selection.

Automatic Redial

Manual Line Selection, Outgoing Line Selection or Prime Line Selection is interpreted as accepting the Automatic Redial (ARDL) by the calling party.

Call Waiting

A call on the Call Waiting key is not selected.

Dial Intercom

A Dial Intercom DN is selected by Incoming Ringing Line Selection and Outgoing Line Selection.

Group Call

This feature is not selected for automatic Outgoing Line Selection or Non-Ringing Line Selection. It is selected for Incoming Ringing Line Selection.

Hot Line

Since the Hot Line key acts as a Single Call Ring (SCR) key, incoming ringing line preference can be applied. Outgoing line preference automatically selects a line other than the current Hot Line, so that a Hot Line call is not accidentally activated

Private Line Service

A Private line DN is selected by Incoming Ringing/Non-Ringing Line Selection and Outgoing Line Selection.

Voice Call

This feature is not selected by automatic Outgoing Line Selection. It is selected for Incoming Ringing and Non-Ringing Line Selection.

Feature packaging

Automatic Line Selection (LSEL) package 72 has no feature package dependencies.

Feature implementation

LD 11 – Assign Automatic Line Selection for each Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(IRD) IRA (NID) NIA (OLD) OLA	(Deny) allow incoming ringing line preference. (Deny) allow incoming non-ringing line preference. (Deny) allow outgoing line preference.
LPK	xx	Specify the last key to be scanned for line preference (such as 0-7, 10-17, 20-27). Prompted only if CLS = IRA, NIA, or OLA. ***Note:*** A value of 0 (zero) for LPK disables this feature.

Feature operation

No specific operating procedures are required to use this feature.

Automatic Number Identification

Contents

This section contains information on the following topics:

Feature description 439
Operating parameters 449
Feature interactions 450
Feature packaging 456
Feature implementation 456
Feature operation 460

Feature description

The Automatic Number Identification (ANI) feature automatically identifies a station originating an outgoing toll call and its destination party and transmits the information to a recording office.

A system with ANI sends information about stations involved in an outgoing toll call, using Multifrequency (MF) signaling, over Central Automatic Message Accounting (CAMA) trunks to toll-switching CAMA, Traffic Operator Position System (TOPS) or Traffic Service Position System (TSPS) offices.

The software portion of ANI performs the following functions:

- identifies an originating outgoing toll call
- determines the calling station identification, and controls the signaling and supervision of the ANI trunk circuit

- connects the MF sender and the ANI trunk circuit
- loads up to 16 digits that are to be MF outpulsed over the ANI trunk into the MF sender
- orders initiation of the outpulsing
- removes the connection between the trunk and the MF sender and establishes the speech path to the trunk

Note: With the E.164/ESN Numbering Plan Expansion, the MF sender card can send 32 digits to the XCT card. This allows an International Number to be sent in one ANI message, instead of two ANI messages.

ANI signaling

E&M, DX or loop signaling sends ANI information to the Central Office. ANI supports three basic methods: Bell, NT400 and NT500.

- The Bell method interfaces the system to
  - — Bell system TOPS, TSPS or CAMA offices
  - — Strowger Automatic Toll Ticketing (SATT) systems types 57, 59, 62, and 70A. These systems accept 1+ and 0+ calls from the system using MF pulsing through customer-provided adapter circuits
  - — Stromberg Carlson Ticketing Systems
- The NT400 method (Modes A and B) is an interface to the Nortel Networks NT400 ticketing system. Mode A repeats the toll access code (0 or 1) in the called number, whereas Mode B does not.
- The NT500 method (Modes A, B and C) interfaces to Nortel Networks NT500 ticketing systems.
  - — Mode A repeats the Access Code (0 or 1) in the called number format for Central Offices that use MF outpulsing and combined trunk groups.
  - — Mode B does not repeat the access code.
  - — Mode C is used in Central Offices with MF outpulsing and trunk groups dedicated only to 1+ or 0+ calls.

The Bell and the NT400/500 methods have different supervisory signals and different number formatting, as illustrated in Figure 6 on page 442 and Figure 7 on page 447. Additionally, there are formatting differences between the NT400 and NT500 method. Tables 18 through 21 summarize the possible combinations of trunk types and ANI signaling methods.

The MF sender cable allows the system to independently outpulse up to 16 digits (including starting and ending digits, called KP and ST respectively) in each of the 30 possible network loop time slots. With the E.164/ESN Numbering Plan Expansion feature, the MF sender can send up to 32 digits. Therefore, an International Number can be sent in one ANI message, instead of two ANI messages.

Figure 6
Supervisory signals (Bell method)

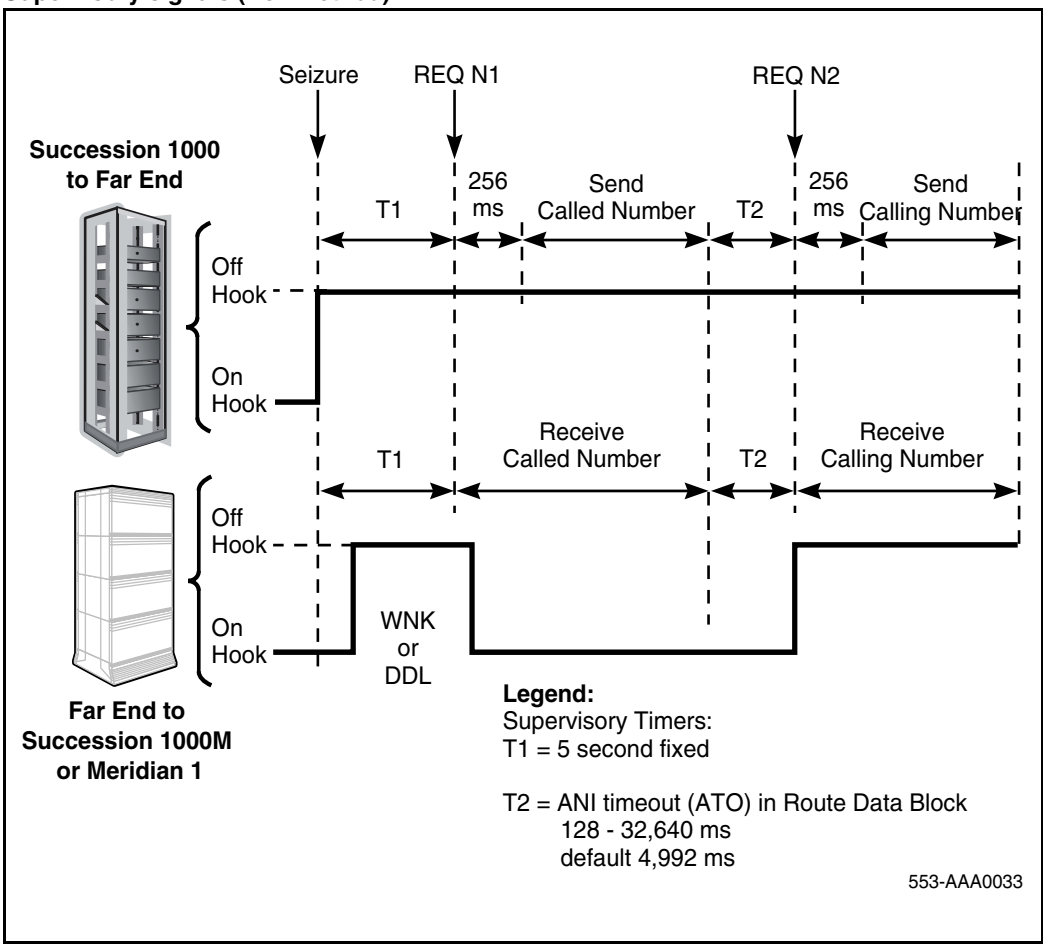


Table 18
Called and calling number information format (Bell method)

Dial Pulse (DP) sending of called numbers			
Call type	**Called number**	**Calling number**	
		Regular trunk group	**Supervisory trunk group**
0	seizure no digits	KP+ID+7D+STP	KP+ID+ST3P
0+7/10D	7/10D	KP+ID+7D+STP	KP+ID+7D+STP
1+7/10D	7/10D	KP+ID+7D+ST	KP+ID+7D+ST2P
011+CC+NN	11+CC+NN	KP+ID+7D+ST	KP+ID+7D+ST2P
01+CC+NN	1+CC+NN	KP+ID+7D+STP	KP+ID+7D+ST3P
010	10	KP+ID+7D+STP	KP+ID+7D+ST3P

Modified Bell Multifrequency sending mode (M2B)			
Call type	**Called number**		**Calling number**
	Regular trunk group	**Super trunk group**	
0	KP+STP	KP+ST3P	KP+ID+7D+ST
0+7/10D	KP+7/10D+STP	KP+7/10D+ST3P	KP+ID+7D+ST
00	KP+0+STP	KP+0+ST3P	KP+ID+7D+ST
00+7/10D	KP+0+7/10D+STP	KP+0+7/10DST3P	KP+ID+7D+ST
1+7/10D	KP+7/10D+ST	KP+7/10D+ST2P	KP+ID+7D+ST
011+CC+NN	KP+1+CC+NN+ST	KP+1+CC+NN+ST2P	KP+ID+7D+ST
01+CC+NN	KP+1+CC+NN+STP	KP+1+CC+NN+ST3P	KP+ID+7D+ST
010	KP+1+STP or KP+10+STP	KP+1+ST3 or KP+10+ST3P	KP+ID+7D+ST or KP+ID+7D+ST

Legend:

0+	=	Operator-assisted call, more digits dialed
0-	=	Operator-assisted call, no other digits dialed
00+	=	Toll operator assisted call, and any other digits dialed
00-	=	Toll operator assisted call, no other digits dialed
1+	=	DDD call
CC	=	Country code
NN	=	National number
ID	=	Information digit
KP	=	Prepare for digits signal
ST	=	End of pulsing
STP	=	Premium
ST2P	=	Identifier error

Table 19
Called and calling number information format (NT400 method)

Mode	Call type	Called number	Calling number
A	0+	KP+0+7/10D+ST	KP+CAT+7D+ST[1]
	0-	KP+0+ST	KP+CAT+7D+ST[1]
	1+	KP+1+7/10D+ST	KP+CAT+7D+ST[1]
B	1+	KP+7/10D+ST	KP+CM+CAT+7D+ST[1]
	0-	KP+ST	KP+CM+CAT+7D+ST[1]
	1+	KP+7/10D+ST	KP+CM+CAT+7D+ST[1]

Legend:

CM	=	1 (for 1+ calls)
	=	STP (for 0± calls)
CAT	=	XX (category digits)
X	=	0, 1,...,9, and XX is customer-defined data defining the type of long-distance call
ST[1]	=	ST (normal)
	=	ST2P (identifier failure)
ST[2]	=	ST2P (identifier error)
	=	KP (station-to-station 1+)
	=	STP (premium 0±)

Table 20
Called and calling number information format (NT500 method)

Mode	Call type	Called number		Calling number
		Dial Pulse (DP) sending	Multifrequency sending	
A	0+	0+7/10D	KP+0+7/10D+ST	KP+CAT+7D+ST[1]
	0-	0	KP+0+ST	KP+CAT+7D+ST[1]
	1+	1+7/10D	KP+1+7/10D+ST	KP+CAT+7D+ST[1]
B	0+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[2]
	0-	not applicable	KP+ST	KP+CAT+7D+ST[2]
	1+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[2]
C	0+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[1]
	0-	not applicable	KP+ST	KP+CAT+7D+ST[1]
	1+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[1]

Legend:

CM = 1 (for 1+ calls)

= STP (for 0± calls)

X = 0, 1,...,9, and XX is customer-defined data defining the type of long-distance call

ST[1] = ST (normal)

= ST2P (identifier failure)

ST[2] = ST2P (identifier error)

= KP (station-to-station 1+)

= STP (premium 0±)

Table 21
Possible combinations of trunk types and ANI methods

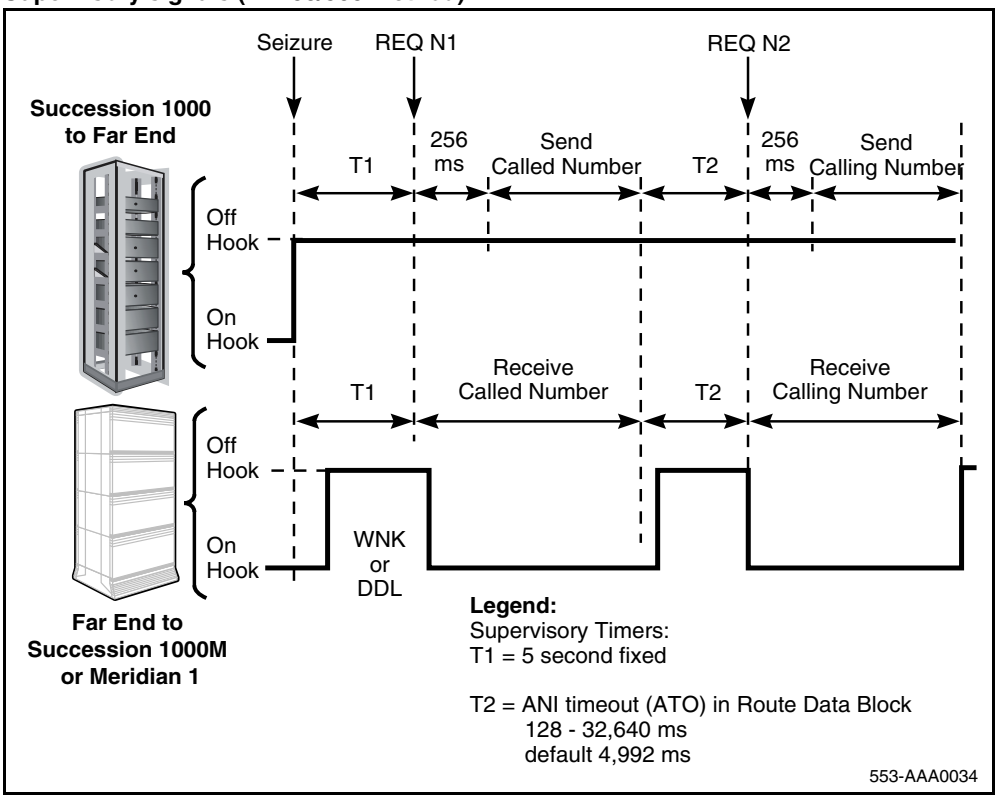
Trunk type	Bell	NT400 A	NT400 B	NT500 A	NT500 B	NT500 C
CAMA-MF	A	A	A	A	A	A
CAMA-DP	A	N	N	A	N	N
CCSA-MF	A	A	A	A	A	A

Legend:

A = Allowed

N = Not allowed

Figure 7
Supervisory signals (NT400/500 method)



Calling and called number information

The called number information always includes the Directory Number (DN) dialed (typically seven or ten digits). The information can also include the toll access code (typically 0 or 1). Multifrequency (MF) sending includes additional control signals such as KP (preparatory digits) or ST (end of pulsing).

The calling number information is always sent in MF. It consists of a calling Directory Number (always seven digits), the preparatory and end-of-pulsing signals and other auxiliary signals. For example, an information digit with the Bell method and class mark and category digits with the NT methods.

Each customer system is assigned a three-, four-, or five-digit Automatic Number Identification (ANI) Listed Directory Number that identifies the customer to the toll office. The calling number for ANI is obtained by combining the ANI LDN with one of the following:

- Analog (500/2500 type) set: Directory Number (DN) of the telephone
- Meridian 1 proprietary telephone: primary DN of the telephone
- Attendant: ANI attendant number specified on a “per customer” basis
- TIE trunk: ANI trunk number specified on a “per trunk group” basis.

The Directory Number Expansion (DNXP) package allows an internal DN to have up to seven digits. If the system is equipped with this package, all DN types listed can be expanded to seven digits maximum. Their combined length with the ANI LDN must remain at seven digits.

The ANI Listed Directory Number is based on the customer’s dialing plan. Otherwise, only the leading digits of a DN (station, attendant or TIE trunk) are retained in the ANI calling number. The full seven digits of a DN can be used as the ANI calling number, provided that no ANI Listed Directory Number is configured.

The calling number information is obtained immediately before being sent. Calls that are modified (For example, calls that are attendant extended or transferred) are billed against the party that initiated the trunk call. (This publication is consistent with Automatic Identification of Outward Dialing).

Automatic Number Identification (ANI)/Central Automatic Message Accounting (CAMA) Enhancement

Two call types allow the ANI Bell method to handle 00- and 00+ calls. Customers dialing 00 can transmit KP + 0 + STP to access toll operator assistance. When 0 is dialed, customers can transmit KP + STP to access local operator assistance. Table 22 on page 449 shows the actions taken by calling 00 and other combinations starting with 0.

Table 22
Actions taken with 00- and 00+ calls

Called number	Bell MF M1A action taken	Bell MF M2B action taken
0	KP + STP	KP + STP
0 + 7/10D	KP + 7/10D + STP	KP + 7/10D + ST3P
00	Overflow	KP + 0 + ST3P
00 + 7/10D	Overflow	KP + 0 + 7/10DST3P

After an ANI/CAMA route has been accessed, the system receive digits representing the called number. Table 22 identifies the actions taken.

> ***Note:*** M1A represents the current Bell MF signaling mode. M2B represents the modified Bell MF signaling mode.

Automatic Number Identification/Central Automatic Message Accounting

CAMA routes using Bell MF signaling Mode B outpulse KP + 0 + …, + START and allow 00- and 00+ calls. 00- and 00+ calls are denied for routes using a different signaling mode.

Controlled Class of Service Allowed (CCSA)

CCSA routes do not support ANI/CAMA.

Route Selection (RS)-Automatic Number Identification

Route Selection for ANI does not support 00- and 00+ dialing. Calls made using 00+ or 00- are treated as 0+ calls. The RS-ANI Data Block determines the 0+ call routing.

Operating parameters

Automatic Number Identification (ANI)/Digital Trunk Interface (DTI) supports CAMA trunks. CCSA-ANI trunks are not supported.

ANI/CAMA operates on a route basis and applies to CAMA routes using the Bell MF signaling method only.

All route members must have a Multifrequency Route (MFR) Class of Service (CLS).

ANI/CAMA is not supported over Dial Pulse trunks. When activating this feature, do not use mixed trunk members.

If 1 or 0 is not dialed following the Trunk Access Code, the system intercepts all outgoing calls over CAMA trunks. This restriction does not apply to outgoing calls over CCSA-ANI trunks.

For E&M or DX signaling, use the QPC71 E&M/DX/Paging trunk circuit card. This card does not have to be modified for ANI. For a complete description of this trunk circuit card.

For loop signaling, the QPC72 loop signaling trunk circuit card is used.

> ***Note:*** The two trunk cards mentioned above provide compatibility with the signaling and supervision requirements of CAMA trunks. They also provide a path for the eventual analog transmission of the MF tones and for speech transmission.

Feature interactions

Directory Number Expansion

If the DN Expansion package is equipped, the ANI billing number (ANAT) can have up to seven digits. The total number of digits for ANAT and ANI listed DN (ANLD) cannot exceed seven.

INIT ACD Queue Call Restore

Restored calls do not retain ANI information, unless the call was an incoming call on an M911 trunk.

M911

The Meridian 911 permits special treatment for emergency calls. This feature requires the QPC916 MF receiver card. For more information about this card, see Feature Group D description and operation.

Valid Automatic Number Identification combinations

When the system receives a call from a 911 trunk, the trunk receives the ANI information through MF signaling from the Central Office. A valid ANI, received through 911, includes a 1-digit NPD or ID digit followed by a 7-digit calling number. The NPD or identification digit can be displayed directly on the answering set display or can be translated to a Numbering Plan Area (NPA) using the Numbering Plan Identification (NPID) translation table in LD 16.

The following are valid ANI digit combinations:

- KP A NXX-XXX ST (where A= the NPD, which can be 0–9);
- KP I NXX XXX ST (where I = an information digit, which can be 0–9;)
- KP I ST (where I = the information digit for ANI failure or Operator Number Identification (ONI). ANI failure is usually designated by a 2 and ONI by a 1); and
- KP A ST (where A denotes maintenance testing, typically the digit 8).

If only one digit is received and that digit is defined in the NPID table as TEST or FAIL, the call is treated as a test case or a call with ANI failure.

Table 23 shows an example of an NPID table. The last two fields, ANI Failure and Test Calls, are mutually exclusive. If the NPD/ID digit 0 is interpreted as ANI failure, it cannot also be interpreted as a test call.

Table 23
Interpreting NPD/ID numbers (Part 1 of 2)

NPI/Info Digit	NPA	ANI Failure	Test Call
0	408	No	No
1	415	No	No
2	NONE	No	No
3	NONE	No	No
4	NONE	No	No
5	NONE	No	No

Table 23
Interpreting NPD/ID numbers (Part 2 of 2)

NPI/Info Digit	NPA	ANI Failure	Test Call
6	NONE	No	No
7	NONE	Yes	No
8	NONE	No	Yes
9	NONE	No	No

If the NPA is not specified (NPA = NONE), the NPD/ID digit appears on the set. Otherwise, the NPA appears on the set for calls with a valid ANI.

Seven zeros indicate a failure (for example, MF receive fault, garbled tones or a timeout). After all ANI digits are received or a timeout occurs, the system processes the call.

A test call has no display.

Trunk route assignments

The 911 trunk must auto-terminate to a Controlled Directory Number (CDN) defined in LD 23. The start arrangement must be WINK and the Class of Service must be defined as Priority Trunk (APY) and Multifrequency Receiver (MFR).

ANI failure

If ANI information is incorrectly delivered, the call may not have a valid ANI, as indicated by the seven zeros in the display.

ANI failure affects the incoming call's Application Module Link (AML) message, which informs the application with a special DN type value. The 911 caller's DN type Information Element (IE) contains one of these types: ANI with NPD, ANI with ID or ANI failure.

Some Central Offices indicate ANI failure with an 8-digit string consisting of NPD followed by 911-0YYY, where YYY denotes the problem. The ANI string 911-0YYY is not treated as a failure so that the digits appear on the screen rather than being overwritten by seven zeros.

Redundancy and call loss requirements are very precise. If the AML terminal display is unavailable (for example, if the host computer is down), the ANI information still appears on the set display.

CDR for 911 ANI calls

If CLID is set to YES in LD 17, 911 ANI information is included in CDR Q records (connection records). CDR records affected are Normal Records, Start/End Records, Authorization Code Records, Connection Records (Q, R, F) and Charge Account Records.

The CDR Q record option is not recommended, since the Meridian 911 application does not need Connection Records and they consume valuable CPU real time. The CDRQ record can nonetheless be configured to include ANI.

Route Selection (RS-ANI)

The optional Route Selection (RS-ANI) is provided with ANI. RS-ANI routes toll calls automatically through specified trunks to toll offices and routes local calls through CO trunks to local switching offices.

To place an outgoing CO call, the station user dials the RS-ANI Access Code (typically 9), followed by a CO Directory Number. If the user dials 0 or 1 after the Access Code, the call routes through a toll trunk group; otherwise, the call routes through a CO trunk group.

Operation

After receiving the RS-ANI Access Code, the system sends the user the second dial tone. The user has 30 seconds to dial a digit or digits. Following this time frame, the system removes the dial tone and provides overflow tone for an additional 15 seconds. The second dial tone is removed after the first digit or digits are dialed. Table 24 shows the system action that corresponds to the digit dialed.

Although it does provide an overflow tone if the user presses the octothorpe key (#), the system ignores the asterisk (*) key. If 0# is dialed, the system activates a 4-second timer and times out.

Table 24
RS-ANI operation

Digit dialed	System action taken
0	A four second timer starts to monitor the next digit dialed. Routing is based on this digit, as follows:
none	The timer times out and the call (0-) routes through the trunk group specified for 0- calls.
1	The timer cancels, and the call (IDD) routes through the trunk group specified for 1+ of IDD calls.
2–9	The timer cancels, and the call (0+) routes through the trunk group specified for 0+ calls.

Trunk types

TIE trunks access RS-ANI as stations do, but all other trunks are intercepted. Any type of trunk can be used for RS-ANI, with the exception of special-purpose trunks such as Paging, Dictation or Recorded Announcement. Normally, the trunk routes shown in Table 25 are used.

Table 25
Trunk route types

Call type	Trunk type
0±	Central Automatic Message Accounting (CAMA)
1+, 011+, 01+, 010-	Central Automatic Message Accounting (CAMA)
other	Central Office (CO)

Class of Service options

Conditionally Unrestricted station Class of Service places non-ARS-handled toll calls through ANI. Refer to *Software Input/Output: Administration* (553-3001-311) to implement this option. See Table 26 for RS-ANI Class of Service options.

Table 26
RS-ANI Class of Service options

Option	Explanation
UNR	Allowed to receive calls from and originate calls to the exchange network (CO, FX, WATS). This includes toll calls.
CUN	UNR for calls placed through ARS and for calls placed through ANI TLD for all other calls
CTD	UNR for calls placed through ARS TLD for all other calls
TLD	Allowed to receive calls from the exchange network; allowed dial access to local exchange network; allowed access to toll network by means of system attendant only; denied access to exchange operator

Legend

UNR = Unrestricted
CUN = Conditionally Unrestricted
CTD = Conditionally Toll Denied
TLD = Toll Restricted Service

New Flexible Code Restriction

Calls from Toll Denied (TLD) stations routed by Automatic Number Identification (ANI) are subject to NFCR. Calls placed by Conditionally Toll Denied (CTD) and Conditionally Unrestricted (CUN) Class of Service stations subject to ANI are treated as unrestricted calls.

Trunk Optimization

ANI trunks allow the Trunk Optimization (TRO) feature to be used whenever calls are routed over PRI and ISL trunks. For additional information on this feature, refer to *ISDN Basic Rate Interface: Features* (553-3001-380).

Feature packaging

Automatic Number Identification (ANI) is package 12. The following packages are also required:

- ANI Route Selection (ANIR) package 13, which requires:
  - — Automatic Number Identification (ANI) package 12

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the ANI customer data.

2 LD 16 – Configure the Centralized Automatic Message Accounting (CAMA) route data.

3 LD 14 – Configure the Centralized Automatic Message Accounting (CAMA) trunk data.

4 LD 28 – Configure the Route selection data for ANI calls.

5 LD 16 – Configure the Centralized Automatic Message Accounting (CAMA) route data.

LD 15 – Configure the ANI customer data.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ANI	Automatic Number Identification.

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
ANAT	xxx...x	ANI billing number for attendants making ANI calls. (The total number of digits in ANAT and ANLD cannot exceed seven digits.)
ANLD	xxx...x	ANI listed DN for billing purposes (0-5 digits). (The total number of digits in ANAT and ANLD cannot exceed seven digits.)

LD 16 – Configure the Centralized Automatic Message Accounting (CAMA) route data.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	CAM CAA	SIGL = Bel, NT4, or NT5. SIGL = Bel.
SIGL	BEL NT4 NT5	Bell method signaling. ITT-North NT400 signaling (only if TKTP = CAM). ITT-North NT500 signaling (only if TKTP = CAM).
FORM	M1A M2B M3C	For BEL, NT4, or NT5 (NT4 and NT5 not applicable if TKTP = CAA). For BEL, NT4, or NT5 (NT4 and NT5 not applicable if TKTP = CAA). For NT5 (only if TKTP = CAM).
ICOG	OGT	Outgoing.
ACOD	xxxx	Access Code.

ID	0-9	Identification digit for CAMA routes (for BEL).
CAT	00-99	Category digits for CAMA routes (only if TKTP = CAM). For NT4 and NT5.
STRK	(NO) YES	(Disable) enable super trunk group feature (Bell method signaling only).
SPTO	(NO) YES	7- to 10-digit, or 3-digit outpulsing for ANI calls.
ANKP	(NO) YES	(Do not) suppress KP signal on ANI calls.
CNTL	(NO) YES	(Do not) allow changes to timers.
- TIMR	ATO 128–65,408	ANI timeout timer in milliseconds (default is 4,992).
- ANDT	(NO) YES	(Do not) provide ANI dial tone.

LD 14 – Configure the Centralized Automatic Message Accounting (CAMA) trunk data.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data. Change existing data.
TYPE	CAM CAA	CAMA trunk. CAMA-ANI trunk (SIGL = BEL in LD 16).
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XUT XEM EXUT	Extended Universal Trunk card. Extended E & M trunk card. Enhanced Extended Universal Trunk.
...		
CUST	xx	Customer number, as defined in LD 15
...		

RTMB	0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
...		
SIGL	DPN DAS	Digital Private Network Signaling System Number 1. Digital Access Signaling System Number 2.
...		
SUPN	(NO) YES	Answer and disconnect supervision required.
...		
CLS	aaa	Class of Service options.

LD 28 – Configure the Route selection data for ANI calls.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RSA	Route selection for ANI.
RASC	xxxx	RS-ANI access code digits.
0-RT	xxxx	Route access code for 0- calls.
0+RT	xxxx	Route access code for 0+ calls.
1RT	xxxx	Route access code for 1+ or IDDD calls.
CORT	xxxx	Route access code for local calls.

LD 16 – Configure the Centralized Automatic Message Accounting (CAMA) route data.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.

TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE CCSA	Trunk type.
ANTK	xxxx	Billing number for TIE or CCSA trunks that are allowed a tandem connection to ANI.

Feature operation

No specific operating procedures are required to use this feature.

Automatic Number Identification on DTI

Contents

This section contains information on the following topics:

Feature description 461
Operating parameters 461
Feature interactions 462
Feature packaging 462
Feature implementation 462
Feature operation 463

Feature description

Automatic Number Identification (ANI) on Digital Trunk Interface (DTI) extends the ANI feature to digital Central Office (DCO) and Digital Toll Office (DTO) trunks. In addition, the ANI capability is extended to Primary Rate Access (PRA) trunk routes through the Primary Rate Interface.

For further information, refer to the Automatic Number Identification feature module in this guide.

Operating parameters

The QPC189F or NT817 (all vintages) are required to support this feature.

DTI interfaces externally with a digital trunk carrier facility at the DS-1 rate. MF signals pass across this interface in a digitally encoded format.

Supervisory signaling through DTI is accomplished by A&B bit signaling. A&B bit signaling can emulate E&M or loop signaling.

Address (called number) signaling through DTI can be dial pulse or MF. Immediate start or wink start may be used.

Calling number information signaling is done using the MF signaling method.

This enhancement supports the three basic signaling methods for ANI. These are Bell, NT400, and NT500.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This enhancement is included in Automatic Number Identification (ANI) package 12.

Feature implementation

LD 16 – Define Central Office or Toll Office port types.

Prompt	Response	Description
REQ	NEW CHG	Add, or change
TYPE	RBD	Route Data Block.
...		
DTRK	(NO) YES	Digital trunk route.
DGTP	DTI	Digital trunk type.
PTYP	(DCO) DTO	CO or Toll Office port type (default DCO).

Feature operation

No specific operating procedures are required to use this feature.

Automatic Preselection of Prime Directory Number

Contents

This section contains information on the following topics:

Feature description 465
Operating parameters 465
Feature interactions 466
Feature packaging 466
Feature implementation 466
Feature operation 467

Feature description

Automatic Preselection allows a user to select the Directory Number (DN) assigned to key zero by lifting the handset. It is not necessary to operate the DN key to get dial tone or to answer an incoming call. The DN assigned to key zero is referred to as the Prime Directory Number (PDN) for that telephone.

Operating parameters

The Automatic Preselection feature does not apply to single-line telephones.

Feature interactions

Automatic Redial

If a call is processed on key 0 and the calling party lifts the handset and selects the Prime Directory Number (PDN), this is interpreted as accepting a redialed call.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Assign PDN to key 0 on proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Telephone type
...		
CLS	CLS	Class of Service options
	(PDN)	Primary Directory Number
...		

KEY	xx aaa yyyy (cccc ***or*** D) zz..z	Telephone function key assignments The following key assignments determine calling options and features available to a telephone. Note that KEY is prompted until just a carriage return <cr> is entered. Where: **xx** = key number 0 **aaa** = SCR, Single Call Ringing **yyyy** = PDN, Primary Directory Number **zz..z** = additional information required for the key aaa. The cccc or D entry deals specifically with the Calling Line identification feature. Where: **cccc** = CLID table entry of (0)-N, where N = the value entered at the SIZE prompt in LD 15 minus 1. **D** = the character “D”. When the character “D” is entered, the system searches the DN keys from key 0 and up, to find a DN key with CLID table entry. The CLID associated with the found DN key will then be used.

Feature operation

With this feature enabled, lifting the handset automatically selects the DN assigned to key zero to receive dial tone or answer an incoming call on that key.

Automatic Redial

Contents

This section contains information on the following topics:

Feature description 469
Operating parameters 471
Feature interactions 472
Feature packaging 476
Feature implementation 476
Feature operation 479

Feature description

Automatic Redial (ARDL) extends the redialing capabilities of the Ring Again and Network Ring Again features. The redialing capabilities of this feature reside at the system level. The system generates redialing attempts that allow the calling party to redial a busy public network subscriber using analog or digital trunks.

This feature is applicable when a calling party dials a public network subscriber number and receives a busy indication. Instead of attempting repeated redial efforts, the calling party can activate ARDL by pressing the Ring Again (RGA) key.

Once activated, the ARDL feature requests the system to automatically redial the attempted dialed number until a successful call termination is completed or until the configured number of redial attempts is reached. A successful call termination is determined when one of the following occurs: a tone detector attached to the call detects a ringback tone, an answer signal is received or an ISDN signaling trunk indicates call termination.

When a successful call termination is detected from the far end, the calling party hears the called party through the set's loudspeaker. The calling party must accept the redialed call within a specified time limit. If not, the redialed call is dropped and not redialed.

Multi-Automatic Redial permits simultaneous activation of the Automatic Redial feature on several RGA keys. This allows more than one number to be redialed in succession. Each Automatic Redial call is attempted once and then another number is attempted. Multi-ARDL numbers are dialed in order of activation.

One set of ARDL calls can be associated with one DN key. Another set of ARDL calls can be associated with a different DN key. This option facilitates the use of the ARDL feature by a secretary who works for several managers. Each manager's DN could be on the secretary's set. A secretary activates the ARDL feature to call different calling groups on both DNs. After a successful call termination, the accepted call is easily accessed by the appropriate manager.

All ARDL requests are associated with the calling party's DN key. Therefore, when the called party is being redialed the calling party's DN key is busy. If the calling party is busy on another DN, the ARDL attempts are redialed on hold. When a successful call termination is completed, the system alerts the calling party by buzzing the set. While ARDL is activated, the calling party's set can be used for incoming/outgoing calls.

ARDL can be activated on a call that has originated from a Single Call Ringing (SCR), Single Call Non Ringing (SCN), Multiple Call Ringing (MCR), Multiple Call Non Ringing (MCN), Private Line key or Hot Line key. The ARDL request is associated with the key from which the call was made. If this key is free, the system attempts to dial the number until a successful call termination is detected and provided a free trunk is available.

Operating parameters

The Ring Again feature must be enabled to operate the ARDL feature.

This feature is only supported on Central Office (CO) and TIE trunks.

The ARDL is supported on proprietary sets, excluding the M3000 and M2317 sets. It is recommended that sets be equipped with display, handsfree and loudspeaker. Analog (500/2500) type sets do not support this feature.

The ARDL feature cannot be activated on data calls.

ARDL can only redial if the Directory Number (DN) key on the calling party's set is idle. For this feature application, only a single external number can be stored against the Ring Again key.

Network Ring Again features do not interfere with the ARDL feature. ARDL is only activated after all Network Ring Again attempts have failed. When ARDL is activated, redial attempts continue with the ARDL feature. ARDL does not support the failure of a DPNSS1 call attempt.

The ARDL feature does not impact the operation of the Ring Again feature on internal calls.

The tone detector is not allocated to detect non-busy tones for off network trunks that have on-board busy tone detectors such as an Extended Flexible Central Office Trunk (XFCOT). Only a busy tone is detected. Accordingly, an Automatic Redial call is considered a successful call even though an overflow tone is sent from the far end.

With the exception of trunks that have on-board busy detectors or an end-to-end Integrated Services Digital Network (ISDN) call, a tone detector is required for all ARDL calls.

If a trunk is not equipped with answer supervision, an ARDL call is redialed once only and then the redial request is cancelled.

The busy tone detector capability is limited to the current tone detector hardware.

This feature introduces the following three timers that control the operation of ARDL:

- The Automatic Redial Acceptance Timer is the maximum allotted time that the calling party has to respond to an ARDL call.

- The Automatic Redial Retry Timer controls the time between successive ARDL retries.
- The Tone Detector Response Timer controls the tone detector response and is defined in LD 16.

Feature interactions

Access Restrictions
Trunk Group Access Restrictions

The Access Restriction/Trunk Group Access Restrictions of an ARDL redialed call are those restrictions that were applied when the call was initiated. These initial restrictions are not changed.

Attendant Barge-In

Attendant Barge In is not allowed to a trunk that is currently used for the ARDL call redialing. This is done to avoid creating a conference when the tone detector is involved.

Attendant Break-In
Attendant Busy Verify

Attendant Break-In and Attendant Busy Verify are not permitted on a proprietary set that is used for an ARDL call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Attendant Blocking of Directory Number

An ARDL redialed call is blocked from the calling party if an attendant uses the Attendant Blocking of Directory Number feature on the calling party's DN.

Attendant Recall
Call Park
Call Transfer
Conference

No Hold Conference
Privacy Release

When an Automatic Redial (ARDL) call is not accepted by the calling party, the following keys are ignored if pressed: Attendant Recall (ARC), Call Park (PRK), Call Transfer (TRN), Conference (A03 or A06), No Hold Conference (NHC) and Privacy Release (PRS).

Autodial

ARDL can be activated on a dialed number using the Autodial (ADL) key.

Automatic Line Selection

Manual Line Selection, Outgoing Line Selection or Prime Line Selection is interpreted as accepting the ARDL by the calling party.

Automatic Preselection of Prime Directory Number

If a call is processed on key 0 and the calling party lifts the handset and selects the Prime Directory Number (PDN), this is interpreted as accepting a redialed call.

Automatic Set Relocation

If the calling party's set is relocated, the ARDL request is cancelled.

Call Detail Recording

The calling party's DN is charged even though a call is not accepted. This occurs because the resources are booked for ARDL attempts.

If Call Detail Recording (CDR) is configured on external calls, additional CDR records are produced. This occurs because each redial attempt produces a CDR record.

Calling Party Privacy

The calling party and called party have the same Calling Party Privacy considerations.

Digit Display

Dialed numbers are displayed when the ARDL feature is activated. The calling party can dial digits even though a busy tone indication is given.

Digits dialed while on hold are not displayed. When the calling party accepts a redialed call, the dialed numbers are displayed. If the Display (DSP) key and appropriate RGA key are pressed while a call is on hold, the number redialed is displayed.

Directory Number - Multiple Appearance

An ARDL call from a Single Call Ringing (SCR) or Single Call Non Ringing (SCN) is only redialed when all sets that have the same DN are free.

An ARDL call from a Multiple Call Ringing (MCR) or Multiple Call Non Ringing (MCN) is only redialed when the originating key is free.

Enhanced Hot Line

An ARDL call can be activated from an Enhanced Hot Line key. However, the call is only redialed when the calling party's HOT key is free.

Last Number Redial

An ARDL call can be activated on a number dialed using the Last Number Redial (LNK) key or by pressing the DN key twice. The ARDL number is saved as the last number redialed.

Line Load Control

ARDL attempts are controlled and restricted by Line Load Control.

Network Alternate Route Selection
Network Speed Call

ARDL can be activated on a Network Alternate Route Selection DN or Network Speed Call.

New Flexible Code Restriction

ARDL calls must pass New Flexible Code Restriction (NFCR) checks. If the redialed number is restricted, the ARDL request is cancelled.

Override

An ARDL call cannot be overridden. This is done to avoid creating a conference when a tone detector is involved.

Pretranslation

ARDL can be activated on a number that has passed the Pretranslation process. However, on an ARDL call the Pretranslation process is not used.

Privacy

If the ARDL call is redialed on a number that is shared with any single line telephone, the ARDL call is accepted when the single line telephone goes off-hook.

Privacy Override

When the Privacy Override feature is activated on the MADN key and the one set activates ARDL, this call can be accepted by other sets.

Private Line Service

An ARDL call can be activated on a Private Line Service key. The call can only be redialed when the calling party's PVR or PVN key is free.

R2 Multifrequency Compelled Signaling

A successful ARDL call dialed through a R2 Multifrequency Compelled Signaling (MFC) trunk is determined by the tone detector (TDET) and MFC. If MFC signaling detects that the call has failed, the ARDL call is cancelled in the same manner as a TDET. If R2 MFC does not detect a call failure a TDET is connected to the call as a regular ARDL call.

Scheduled Access Restrictions

The Scheduled Access Restrictions (SAR) on ARDL redialed calls are set when the call is initiated. If restrictions are changed later, the prior restrictions still apply.

Speed Call
System Speed Call
Stored Number Redial

The Automatic Redial (ARDL) feature can be activated on a call using Speed Call (SCL), System Speed Call (SSU/SSC) or Stored Number Redial (RDL) keys.

Speed Call on Private Lines

The ARDL feature is activated on a number dialed using the Private Line (PVR/PVN) key and then making a speed call by pressing the Speed Call (SCL) key.

Feature packaging

Automatic Redial (ARDL) requires the following packages:

- Automatic Redial (ARDL) package 304
- Ring Again (RGA) package 1
- Tone Detector (TDET) package 65

Outpulsing of Asterisk and Octothorpe (OPAO) package 104 and Automatic Redial (ARDL) package 304 are mutually exclusive. The ARDL package is turned off automatically if both packages are equipped.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 13 – Define Tone Detector Units.

2 LD 15 – Define Automatic Redial.

3 LD 16 – Define Automatic Redial Tone Detector Response Timer.

4 LD 87– Define Automatic Redial Network Route Selection.

5 LD 11 – Assign Automatic Redial Class of Service and Key.

LD 13 – Define Tone Detector Units.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	TDET	Tone Detector data block.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 15 – Define Automatic Redial.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Change features and options.
CUST	0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- ARDL_ATTEMPT	1-(30)-60	Number of Automatic Redial attempts.
REQ:	CHG	Change.
TYPE:	TIM	Change Timers.
CUST	0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

- ARDL_ACCEPT	0-(20)-60	Automatic Redial Acceptance Timer in seconds. Odd number entries are rounded up to the next even number and echoed back with a message.
- ARDL_RETRY	10-(30)-60	Automatic Redial Retry Timer in seconds. Odd number entries are rounded up to the next even number and echoed back with a message.

LD 16 – Define Automatic Redial Tone Detector Response Timer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
...		
CNTL	YES	Changes to controls or timers.
TIMR	RTD 0-(12)-60	Tone Detector Response Timer in seconds. Odd number entries are rounded up to the next even number.

LD 87– Define Automatic Redial Network Route Selection.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
CUST	xx	Customer number, as defined in LD 15
FEAT	NCTL	Network Control Feature.
...		

NCOS	(0) - 99	Network Class of Service group number.
- ARDL	(A) I	A = Automatic Redial network route selection allowed from all route sets (initial and extended). I = Automatic Redial network route selection allowed from initial set of routes only.

LD 11 – Assign Automatic Redial Class of Service and Key.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2016, 2216, or 2616.
...		
CLS	RDLA	Automatic Redial allowed (default). RDLD = Automatic Redial denied.
KEY	xx RGA	Ring Again key assignment.
KEY	xx RGA	Ring Again key assignment for Multi-Automatic Redial capability.

Feature operation

Enable Automatic Redial

1. Press an idle DN key, handsfree key or go off-hook. Dial desired public network number. The DN lamp is lit.
2. The calling party hears busy, overflow, ringback tone, etc.
3. Press the Ring Again (RGA) key.

— For a non-ISDN call, the RGA key lamps lights, the tone stops and the DN lamp goes dark.

— For ISDN call, the RGA key lamps initially flashes while the system attempts to activate the Network Ring Again features. If these features cannot be activated, the RGA lamp is steadily lit and the ARDL feature is activated.

Automatic Redialing - External number

After ARDL is activated the following possibilities can occur:

1 The call is answered by the called party and accepted by the calling party. In this case, both parties are connected.

2 The call is answered by the called party, but the calling party does not accept the call within the predefined time limit. The call is dropped and not redialed again.

3 The called party's number is occupied. The call is dropped and redialed later according to the timer configuration.

4 The call is blocked for some reason such as trunks or network congestion. The call is dropped and redialed later according to the timer configuration.

If the calling party presses another DN key while ARDL is activated, the ARDL attempt continues redialed on hold. If the Release (RLS) key is pressed when an ARDL call is attempted and is the active call, the redial attempt is dropped and redialed later after the predefined allotted time.

Automatic Redial - Originating telephone is idle on another number

1 The calling party's DN is idle.

2 The RGA key lamp winks. The DN lamp is lit and the called number appears on the set's display. A connection is made to the loudspeaker of the calling party's set.

3 One of the following occurs:

— If the called party's number is busy, the call is dropped when a busy tone is detected. The DN goes dark, the RGA lamp is steadily lit and the loudspeaker is deactivated.

— Otherwise, the calling party hears ringback or answer through the loudspeaker.

4 If the calling party accepts the call by going off-hook, pressing the DN key, or pressing the handsfree, both parties are connected and the RGA lamp goes dark.

5 If the calling party does not accept the call within the predefined time limit, the DN key goes dark and one of the following occurs:

— If the called party answers the call, the following occurs: the RGA key goes dark, the redial call is disconnected and the ARDL request is cancelled. The display returns to an idle state.

— If the called party does not answer the call, the following occurs: the call is disconnected and the RGA lamp is steadily lit. The ARDL request is not cancelled and is ready for another redial attempt. The display returns to an idle state.

Accept an Automatic Redial call

To accept an ARDL call, the calling party performs one of the following:

1 If the ARDL call is the active call on the set, then the calling party must lift handset, press handsfree or press the key on which the ARDL call is active.

2 If the ARDL call is dialed on hold the calling party must press the associated DN key.

Cancel Automatic Redial

Automatic Recall is canceled in the following cases:

- The calling party presses the lit or winking RGA key (the cancellation request can be activated between ARDL attempts or during a redial attempt).
- The calling party accepts the ARDL call.

- The ARDL call is redialed the predefined number of call retries.
- The calling party does not accept a successful ARDL call within the predefined time limit.

At cancellation of the ARDL request, the RGA key lamp goes dark.

Automatic Set Relocation

Contents

This section contains information on the following topics:

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Feature operation 493

Feature description

Automatic Set Relocation (ASR) and Modular Telephone Relocation (MTR) move a telephone to another location without the intervention of a craftsperson. MTR reduces the number of steps required to relocate the Meridian Modular Terminals.

With ASR, Directory Numbers (DNs) and features assigned to the telephone are maintained. Up to 32 telephones can be relocated at any one time. The following access codes are associated with this feature:

- Special Prefix code (SPRE) relocation code 81
  SPRE codes are system codes enabling analog (500/2500 type) telephones to utilize additional telephone features. Refer to the Telephones feature module in this guide.

- Flexible Feature Code (FFC) relocation number
  FFCs are user programmable codes that enable analog (500/2500 type) telephones to access certain telephone features. Refer to the Flexible Feature Code feature module in this guide.
- Security code
  You must enter the security code before a telephone can be moved.
- Identification code
  The identification code is user selectable, and can be any four-digit number (excluding the symbols * and #). (MTR does not require this code.)

This feature is also used to install and enable line cards to make unused telephone locations available for telephone relocation. Adding the first telephone on a line card by using the Service Change overlay enables that card (if it is not already enabled). Removing the last telephone from a line card leaves that card enabled; it does not disable the card.

Automatic Set Relocation (ASR) requires the circuit units on SL-1 and digital line cards used for supplementary power to be specified as power units in LD 12. This allows the system to disable signaling to these units, while leaving unequipped units enabled for telephone relocation. If power units are not specified, they generate erroneous messages and may disable the entire card.

After putting a telephone back into service, the craftsperson should wait at least 20 seconds before using the telephone.

Modular Telephone Relocation

Modular Telephone Relocation enhances ASR to make relocating Meridian Modular Telephones simpler and faster (by omitting the requirement for an identification code). The following telephones support Modular Terminal Relocation:

- M2006
- M2008
- M2016S

- M2216
- M2616

When a telephone is relocated out, a relocation block is automatically built to store the relocation information in the protected data area. The relocation block includes the old Terminal Number (TN), the terminal ID information, the serial number of the telephone, and feature information. If a data dump occurs, the relocation block is not copied to the disk.

Modular Telephone Relocation uses the unique serial number and terminal ID of the Meridian Modular Telephones (instead of the identification code) to identify the one being relocated. This reduces the number of steps needed for relocation.

A telephone's successful relocation is indicated by a 180-millisecond buzz through the telephone's loudspeaker, not a tone through the handset. The buzz occurs after the telephone is plugged into the new location, and the parameter download to the Meridian Modular Terminal is complete.

Modify the relocation table

The relocation table contains information regarding the telephone's serial number, Terminal Number, and terminal identification. When a telephone is relocated OUT, the table maintains the necessary telephone information. When the telephone is relocated IN, the system searches the table for that telephone's information. When the information is found, the data is moved to the new location. The telephone data is then removed from the relocation table.

Through LD 50, the serial number or any terminal ID information may be modified while the telephone is relocated out (before it has been relocated back in). For example, use LD 50 when replacing a telephone with another one of the same type with a different serial number or terminal ID, but the same key configuration.

LD 21 prints information about telephones that have been relocated out.

The IDU (ID for Unit) command in LD 32 determines the telephone's serial number and ID information.

Operating parameters

A single-line telephone must be relocated to a vacant position on an analog (500/2500 type) Line Card.

A digital telephone must be relocated to a vacant position on a Digital Line Card (DLC) or Integrated Services Digital Line Card (ISDLC) in the switch.

Moving a telephone from an off-premise to on-premise location or vice versa is not recommended, as incorrect pad values on connections may result.

A Manual Line telephone cannot be relocated using the Automatic Set Relocation feature.

The relocation table allows a maximum of 32 telephones to be relocated out at one time.

A relocated out telephone cannot be relocated in to an already defined TN. A telephone being relocated in must be plugged into a TN location that currently has no assigned telephone information.

Automatic Call Distribution (ACD) agent telephones with an associated supervisor and the ACD supervisor telephones cannot be relocated.

If a data dump occurs while a telephone is relocated out, a SYSLOAD returns the telephone to its original TN location. If a telephone was in the relocated out state when the last data dump occurred, and has since relocated in, another data dump is necessary. The second data dump prevents a SYSLOAD from returning the telephone to its previous TN location.

When Modular Telephone Relocation is used and the overflow tone is returned during relocation out, the relocation attempt is abandoned. Try the relocation again.

When Modular Telephone Relocation is used, there is a slight delay between the time the telephone is plugged in and the buzz. The buzz occurs after the telephone is relocated in, enabled, and downloaded. This delay is traffic dependent. If no buzz is received, the relocation is unsuccessful.

When Modular Telephone Relocation is used and a telephone is relocated out, a Customer Service Change (CSC) message containing the old TN number, serial number, and terminal ID is displayed on the TTY. When a telephone is relocated in, a CSC message containing the old TN and new TN is displayed. These messages are placed in the History File.

When Modular Telephone Relocation is used and a SYSLOAD occurs before a data dump completes, the data for all telephones relocated in or out is lost. Return the telephones to their original location and repeat the relocation process.

Feature interactions

Automatic Redial

If the calling party's set is relocated, the Automatic Redial request is cancelled.

Call Forward No Answer Hunting

Calls will not hunt or forward no answer to a telephone that is being relocated.

Call Forward Ring Again

If Call Forward, or Ring Again is active when a telephone is relocated, the feature is deactivated.

China – Flexible Feature Codes - Busy Number Redial

Enhanced Flexible Feature Codes - Busy Number Redial

Busy Number Redial is deactivated when a set is relocated.

Hunting

Calls will not hunt to a telephone that is being relocated

Make Set Busy

If Make Set Busy is active when the telephone is relocated, Make Set Busy remains active.

Meridian Mail Voice Mailbox Administration

Relocating a user with an associated VMB to a new TN will not affect the VMB. The VMB remains logged in and continues to receive incoming voice messages while the telephone is being relocated.

A telephone that is relocated out but not relocated back in can still have an active VMB. A relocated telephone must be deleted manually on the system before its associated VMB is removed.

Multiple Appearance DN Redirection Prime

The original Multiple Appearance Directory Number Redirection Prime (MARP) TN is restored when the telephone relocates.

When Automatic Set Relocation or Meridian Modular Terminal is used to move a telephone, the telephone's MARP designations are maintained. If the TN is a MARP for one or more DNs, the system maintains the MARP TN. A system message indicates the telephone relocation.

When a set leaves the system due to set relocation, the following CSC message appears:

CSC010 x y
x = old TN (l s c u) for the telephone
y = ID code entered

While the telephone is being relocated, a temporary MARP TN is assigned. The following SCH message appears for each DN associated to the removed MARP TN.

SCH5524 DN nnnn NEW MARP l s c u
nnnn = the DN associated with the MARP TN
l s c u = the new default MARP for DN nnnn

The same message given through Attendant Administration displays on the Attendant Console when a MARP is assigned for a DN. The History File can be configured to store these messages until a printout is requested.

When a telephone reenters the system, the following message appears:

CSC011 x y
x = old TN (l s c u) for the telephone
y = new TN (l s c u) for the telephone

The following message appears again for *each* changed TN:

SCH5524 DN nnnn NEW MARP l s c u
nnnn = the DN associated with the MARP TN
l s c u = the new MARP TN assigned to DN nnnn

Night Key for Direct Inward Dialing Digit Manipulation

Delete the DRC key from a telephone before performing Automatic Set Relocation. If this is not done, the DRC lamp is activated on the wrong telephone.

Power Fail Transfer

Since Power Fail Transfer is hardwired to certain Terminal Numbers, this feature is not maintained by a telephone when it is relocated.

Feature packaging

Automatic Set Relocation (ASR) package 53 has no feature package dependencies.

Modular Telephone Relocation requires the following:

- Automatic Set Relocation (ASR) package 53
- Meridian Modular Terminals (ARIE) package 170
- Digital telephones (DSET) package 88

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Assign the Automatic Set Relocation security code.

2 LD 10 – Enable/disable line circuits for Automatic Set Relocation.

3 LD 11 – Enable/disable line circuits for Automatic Set Relocation.

4 LD 12 – Gather data for each line circuit to be used as a supplementary power source.

5 LD 17 – Allow ASR messages to be printed at a system terminal or stored in the History File.

6 LD 17 – Allow Automatic Set Relocation messages to be printed at a system terminal or stored in the History File.

7 LD 32 – Query information regarding a terminal's type, NT code, color, release number, and unique serial number. This command works only for Meridian Modular Terminals.

8 LD 50 – Remove an entry in the relocation table.

9 LD 21 – Print information in the relocation table.

LD 15 – Assign the Automatic Set Relocation security code.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- SRCD	xxxx <CR> X	Automatic Set Relocation security code; default is 0000; X removes security code.

LD 10 – Enable/disable line circuits for Automatic Set Relocation.

Prompt	Response	Description
REQ:	new out	Configure Automatic Set Relocation.
TYPE:	cardslt	500/2500 line circuit for Automatic Set Relocation.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 11 – Enable/disable line circuits for Automatic Set Relocation.

Prompt	Response	Description
REQ:	new	Add new data.

TYPE:	cardmlt	SL-1 or digital line circuit for Automatic Set Relocation.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 12 – Gather data for each line circuit to be used as a supplementary power source.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PWR	SL-1 line circuit for supplementary power.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 17 – Allow ASR messages to be printed at a system terminal or stored in the History File.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
IOTB	(NO) YES	(Do not) change input/output terminals or devices.
HIST	(0)-65534	History File buffer length.
- ADAN	NEW CHG aaa x	System terminal device number for Automatic Set Relocation messages. aaa and x = HST. PRT 0-15. TTY 0-15.
- USER	CSC	Customer service change (Automatic Set Relocation) messages.

LD 17 – Allow Automatic Set Relocation messages to be printed at a system terminal or stored in the History File.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW CHG aaa x	System terminal device number for Automatic Set Relocation messages. aaa and x = HST. PRT 0-15. TTY 0-15.
- CTYP	aaaa	Card type, where: aaaa = DCHI, MSDL, MSPS, SDI, SDI2, SDI4, or XSDI.
- DNUM	(0-15)	Device number printed automatically (same as ADAN number).
- USER	CSC	Customer service change (Automatic Set Relocation) messages.
CUST	xx	Customer number, as defined in LD 15

LD 32 – Query information regarding a terminal's type, NT code, color, release number, and unique serial number. This command works only for Meridian Modular Terminals.

IDU l s c u		Prints telephone's information.

LD 50 – Remove an entry in the relocation table.

Prompt	Response	Description
REQ	OUT CHG	Remove, or change an entry in the relocation table.
TYPE	MTRT	Modular Telephone Relocation Table.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

SER	xxxxxx	Serial number (prompted for changes only).
NTCD	xxxxxxx	NT code (for changes only).
COLR	xx	Color (prompted for changes only).
RLS	xx	Release (prompted for changes only).

LD 21 – Print information in the relocation table.

Prompt	Response	Description
REQ:	PRT	Print.
TYPE:	SRDT	Set relocation data.

Feature operation

To use Automatic Set Relocation:

1. Lift the handset.
2. Enter the relocation code (either SPRE 81 or the Flexible Feature Code).
3. Enter the security code. The default is 0000.
4. Enter the four-digit code to identify your telephone. A tone confirms the telephone is ready to be moved.
5. Unplug the telephone and install it at the new location.
6. Wait 30 seconds after plugging the set into the new location, lift the handset, and dial the four-digit identifier. A tone confirms the telephone has been moved successfully.

Modular Telephone Relocation

To relocate a telephone using Modular Telephone Relocation:

1. Lift the handset or activate handsfree.
2. Enter the relocation code (either SPRE 81 or the Flexible Feature Code).
3. Enter the security code. The default is 0000.

4 A two-second tone burst confirms that the telephone is relocated out.

5 Unplug the telephone and install it at the new location.

6 The confirmation buzz through the telephone's loudspeaker indicates the telephone is in service.

Note: All calls associated with the telephone receive force disconnect while it is relocated out. The telephone information automatically moves to the relocation table.

Automatic Timed Reminders

Contents

This section contains information on the following topics:

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Operating parameters 495
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Feature implementation 497
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Feature description

Automatic Timed Reminders alert the attendant when a call extended to a station by the Attendant Console has not been answered within a predefined period of time. Recall timers for different conditions can be specified by the customer as follows:

- Slow Answer (set in increments of six seconds)
- Camp-On (set in increments of two seconds)
- Call Waiting (set in increments of two seconds)

If no entry is made, the default is 30 seconds in each case.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Overflow Position

After an attendant call has been rerouted using the Attendant Overflow Position feature, there is no automatic timed recall to the attendant or any other DN.

Call Forward by Call Type

Calls eligible for Flexible Call Forward No Answer treatment, and handled by Call Forward by Call Type, use the Call Forward No Answer timer in the Customer Data Block as the recall timer for attendant extended calls. Irrespective of the relative timeout for Automatic Timed Recall, the ringing continues as long as allowed by the Call Forward No Answer Timer.

Call Forward No Answer
Call Forward No Answer Second Level

When Call Forward No Answer is activated on a telephone, the slow answer timer begins only after the call reaches its final destination.

Call Park

A Call Park recall to an attendant appears on the Recall Incoming Call Indicator.

Call Waiting Redirection

When Call Forward No Answer (CFNA) is active, the Slow Answer Recall timer begins only after the call reaches its final destination. CFNA has precedence over Attendant Recall for attendant-extended calls. Irrespective of the relative time-out intervals for each feature, ringing continues as long as allowed by CFNA for sets with CFNA enabled.

Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also takes precedence over the Call Waiting recall timer.

Directory Number Delayed Ringing (DNDR)

If a dialed set has DNDR defined, and an attendant re-extends a call without releasing it, the DNDR timing is not reset. If the value of the recall timer is less than that of the DNDR timer, the call is recalled to the attendant before audible notification begins.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Define Recall timers and add/change a Recall Incoming Call Indicator key on Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	TIM	Timers.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- RTIM	xxxx yyyy zzzz	Recall timers. xxxx = slow answer, 0-3,066, in six-second increments (default 30 seconds). yyyy = Camp-on, 0-1,022, in two-second increments (default 30 seconds). zzzz = Call Waiting, 0-1,022, in two-second (increments (default 30 seconds).
TYPE	ATT	Attendant Console options.
- ICI	0-19 RLL	Add RECALL ICI to all consoles.

Feature operation

One optional Recall Incoming Call Indicator (ICI) key is provided on the Attendant Console for operator-extended recalls.

Automatic Wake Up

Contents

This section contains information on the following topics:

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Feature operation 517

Feature description

Automatic Wake Up (AWU) provides an efficient wake up service for hospitality and health care environments. It relieves the attendant from having to make wake up calls by providing this service automatically. At the requested time, the system automatically rings the room or extension and connects the called party upon answer to music followed by a recorded wake up announcement.

If the wake up call is answered within a customer-specified number of rings (two to five rings), the system recognizes a completed call and presents the predefined wake up treatment. The system disconnects the AWU call when the called party releases, or when the recording cycle is completed.

The wake up message runs continuously. Upon answering a wake up call, the called party hears music until the message begins again. If the message is 15 seconds long, and the wake up call is answered on the 14th second of the message, the calling party hears one second of music before the message. If the call is answered on the third second of the message, the calling party hears 12 seconds of music first.

The system allows for an alternate recording that can be used for evening wake up calls or when the primary recording is being updated. The secondary recording can also replace the primary recording at a customer-specified time period.

Answer the wake up call

The Wake Up indicator goes dark after the guest answers the wake up call. Customers can set the attendant recall option if the call is unanswered after a specified number of tries (from one to three).

Answering the wake up call for multiple appearance DN telephones is similar to single appearance DN telephones: after the call is answered, the Wake Up indicator goes dark.

The system balances the wake up load over five-minute intervals, generating a maximum of 100 wake up calls per five-minute period. The system processes one wake up call every two seconds during peak periods, and one wake up call every four seconds during lighter periods. A light load is defined as anything less than 60 wake up call requests per five-minute interval.

A wake up request is rejected by the system under the following conditions:

- The wake up request (in units of five-minute intervals) is less than one interval ahead of the current time interval (see Note below).
- The wake up request (in units of five-minute intervals) is less than five intervals before the current time interval. In other words, the wake up request is more than 23 hours and eight intervals in advance.
- The interval requested contains 500 calls already.

Note: The time interval = (hour x 12) + (minute / 5). Always round down to the nearest five-minute interval.

If the interval requested for a wake-up call already contains the maximum number of calls, the system searches for the next available time interval in the following sequence:

- the five-minute interval before the requested time
- the five-minute interval after the requested time
- the next available five-minute interval within three hours before the requested time

You can also use a Background Terminal (BGD) to enter Automatic Wake Up information. The Background Terminal lets you monitor system operation. One or more terminals can be assigned to access AWU data. You can have data displayed or printed at a preselected time of day.

500 Wake Up calls

The number of Automatic Wake Up calls available per five-minute period is 500 calls.

You can define the number of rings for the call from two to five. If there is no answer after the specified number of rings, the AWU call overflows to the next five-minute interval. The system tries three times to terminate the call before it is recalled to the attendant. You can define the number of wake up attempts, from one to three.

No more than 25 analog (500/2500 type) telephones should be ringing at any one time. To ensure this, set the Number of Rings for Wake Up (NRWU) prompt in LD 15 according to the recommendations listed in Table 27 on page 502. The NRWU is two to five, with a default of five.

Table 27
Recommended number of rings per Automatic Wake Up call

Time on (seconds)	Time off (seconds)	Maximum number of rings
2*	4*	5*
3	3	2
2	1	5
1	2	5

* North American standards.

Only 500 AWU calls can be defined for the system, but up to 750 calls can actually be placed. Up to half of the programmed AWU calls unanswered can be carried over to the next five-minute interval. The carry-over from one block to the next is important in limiting the number of calls in the original programmed interval.

For a complete description on programming AWU with the Background Terminal, please refer to *Hospitality Features: Description and Operation* (553-3001-353).

Guest Entry of Auto Wake Up (GEWU) Calls

GEWU provides entry of a wake up call from a room telephone. By using the Wake Up key (WUK) on the telephone, guests can program, query (with display), or cancel their own wake up calls based on a 24-hour time format.

Requests must be made on a daily basis since the wake up time is automatically canceled after each use.

GEWU does not alter the operation of AWU, but adds a new option to AWU programming. Unless otherwise specified, operating GEWU is the same regardless of whether the telephone has a display. The distinction is that with a display, guests can check their wake up call requests. A dash (–) indicates that no time has been programmed. In addition, when programming a wake up call, the system will search for and display the next available time if the time interval chosen for the wake up call is full. Without a display, the guest can still program and cancel a wake up call.

Note: For Multiple Appearance DN telephones, the wake up time for secondary DNs cannot be queried.

Multi-Language Wake Up (MLWU) Calls

MLWU provides Automatic Wake Up calls in any of up to six languages. You can use any language as long as you have a recording of it available on a Recorded Announcement (RAN) trunk.

At check-in, each guest can choose the language for wake up calls. If no language is assigned, the default language, Language 0, is used.

You can assign a language to a room's telephone at any time by using the Background Terminal (BGD) or Property Management System (PMS). A room DN is valid if it has at least one appearance as a Prime DN (key 0) on a telephone and Controlled Class of Service Allowed (CCSA). Multiple appearance telephones with the same Prime DN may be assigned different languages through Service Change.

You can also assign the language on a TN basis, allowing the language option to be employed outside the hospitality industry without requiring a BGD terminal or the PMS. Refer to LD 10 and LD 11 in *ISDN Basic Rate Interface: Installation and Configuration* (553-3001-218) for the "LANG" prompt.

The language remains unchanged until the next language assignment. An AWU language cannot be changed on a call-by-call basis. The customer may, however, optionally clear the language either at check-in or check-out times, using the Background Terminal.

If Automatic Wake Up is enabled, up to six pairs of language-specific RAN routes (both a.m. and p.m. for each language), called Automatic Wake Up routes (AWR), can be configured. The languages, 0-5, correspond to the AWR routes RAN1/RAN2 (for Language 0), LA11/LA12 (for Language 1), up to LA51/LA52 (for Language 5) in the Customer Data Block (LD 15). The only requirement is that the default language routes RAN1 and RAN2 for Language 0 must be defined. If a specific language AWR is not accessible at wake up time, the corresponding primary or secondary default language routes (RAN1 and RAN2) are used.

On a Background Terminal, a customer can define a two-character language identifier to reference the languages. For example, the customer may define Language 0 as EN (English), Language 1 as SP (Spanish), and Language 2 as GR (German). For details on implementing BGD terminal commands, refer to *Hospitality Features: Description and Operation* (553-3001-353).

Unanswered Automatic Wake Up calls recall to the attendant if the attendant recall option is on. Upon a recall, the room's language is displayed on the Attendant Console. On alphanumeric displays (M1250 or M2250 Attendant Consoles), the language identifier is displayed after the Call Party Name Display (CPND) fields.

Multiple Wake Up Flexible Feature Codes

Multiple Wake Up allows up to four wake-up calls to be entered using a Flexible Feature Code (FFC), and allows those calls to be repeated daily, if desired, by entering a separate FFC. The time is in a four-digit 24-hour format (H1 H2 M1 M2). To activate Repeat Multiple Wake Up, the user dials "MWRA H1 H2 M1 M2".

> ***Note:*** If a wake-up time has already been entered using the standard Automatic Wake Up Activate (AWUA) FFC, only three other multiple wake-up times may be entered.

To deactivate a single wake-up time, the user enters "MWUD H1 H2 M1 M2", where MWUD is the Multiple Wake Up Deactivate FFC. To deactivate all wake-up times, the user enters "MWUD#". The general Deactivate (DEAF) FFC does not apply to Multiple Wake Up.

If the MWUD FFC is entered again after all wake-up times have been deactivated, confirmation tone is given. If the MWUD FFC is entered again to deactivate a wake-up time that has been already deactivated, overflow tone is given. If an attempt is made to enter an existing wake-up time, confirmation tone is given. If an attempt is made to enter an existing wake-up time as a repeat wake-up time, then that time is activated as a repeat wake-up time. If an attempt is made to enter an existing repeat wake-up time as a single wake-up time, then that time is activated as a single wake-up time. In both cases, confirmation tone is given.

To verify a Multiple Wake Up time, the user dials "AWUV H1 H2 M1 M2" (where AWUV is the existing Verify Automatic Wake Up FFC).

Operating parameters

To operate AWU, a system must have a Background Terminal or Attendant Console with AWU key, room telephones with Controlled Class of Service Allowed (CCSA), and Recorded Announcement (RAN) trunks.

This feature requires a Background Terminal (BGD).

Each Automatic Wake route requires a minimum of two trunks.

The following hardware is required for the AWU feature:

- QPC74 RAN trunk interface card or NT8D14AH universal trunk card
- a continuous announcement (RAN) machine, such as the Audichron HQ-1 112

  Note: A dedicated conference loop is no longer required for the network-enhanced machines.

For the call to utilize both music and a wake up announcement, an AWR route must be installed and the route must be programmed at the RANF prompt in LD 15. The music source can be wired into the audio pairs of the RAN trunk, or music can be recorded on the RAN device.

Automatic Wake Up is only allowed on a telephone's Prime Directory Number (PDN). For telephones in a multiple-appearance arrangement, all telephones are rung; however, only one wake up time can be assigned against the PDN. The system tries the wake up call a customer-defined number of times (from one to three), and then treats it as any other unanswered wake up call. In a single-call arrangement, if any appearance of the DN is busy when the wake up call is made, the wake up call is not presented. In a multiple-call arrangement, the wake up call is presented to all idle appearances.

A wake up key cannot be configured on a data station (a telephone with DTA Class of Service).

There can only be one wake up key per telephone.

Only Attendant Consoles can have an AWU key. The AWU time to be programmed on digital telephones (using GEWU and a Wake Up key).

Automatic Wake Up and Centralized Attendant Services (CAS) are mutually exclusive.

If the wake up call goes unanswered, or the guest hangs up before the AWU two-second hold time, the system tries the wake up call again in the next five-minute interval. If Attendant recall is enabled, the call transfers to the attendant following the last unsuccessful wake up call attempt.

Maintenance technicians can access any AWU RAN trunk or music trunk with the RAN trunk access code.

For Multiple Wake Up, the FFCs selected must be unique numbers up to seven digits long. They cannot conflict with any DN already in the dialing plan.

The following are not supported for Multiple Wake Up:

- The attendant query for the Multiple Wake Up time
- Multiple Wake Up from attendant administration
- The Background Terminal, Background Terminal Display for Multiple Wake Up
- Traffic for Multiple Wake Up

The Deactivate (DEAF) FFC is not supported for Multiple Wake Up.

Multiple Wake Up is supported only on analog (500/2500 type) telephones.

The Automatic Wake Up feature can be active at the same time as Multiple Wake Up.

If one Automatic Wake Up time has been set using the Automatic Wake Up Activate (AWUA) FFC, only three more Multiple Wake Up calls can be entered using the MWUA FFC.

Feature interactions

Attendant Administration

The Attendant Administration feature does not support data entry or changes for the AWU feature.

Attendant Overflow Position

AWU recalls are not redirected to a customer-defined Attendant Overflow Position DN. Failed wake up calls stay in the attendant queue or ring indefinitely on the console.

Call Party Name Display

All display information associated with Automatic Wake Up (AWU) programming is directed to line three of the display. Names are appended to DNs appearing on line three if they are different from those on line two, or if no DN appears on line two. There is no DN information on line two if the attendant has initiated the AWU process while not on an active call. No DES information is appended, since AWU operates on a DN basis.

Coordinated Dialing Plan

AWU supports Coordinated Dialing Plan as long as an internal DN is used.

Directory Number Delayed Ringing

The Directory Number Delayed Ringing feature is not supported.

Do Not Disturb

When a telephone is configured for Do Not Disturb, a wake up call can still be presented.

Flexible Feature Codes Enhancement

Telephones can activate Automatic Wake Up (AWU) features for their own station with Common Controlled Switching Arrangement Class of Service.

The Automatic Wake Up feature can be active at the same time as Multiple Wake Up.

The attendant query function is not supported for Multiple Wake Up.

Multiple Wake Up from Attendant Consoles is not supported.

The Background Terminal (BGT) is not supported for Multiple Wake Up.

If one Automatic Wake Up time has been set using the Automatic Wake Up Activate (AWUA) FFC, only three additional Multiple Wake Up calls can be entered using the Multiple Wake Up Activate (MWUA) FFC.

Intercept Computer Dial from Directory - Post-dial Operation

This feature can be requested as follows:

- Press the Wake-up key on the Attendant Console.
- Dial a DN from the Intercept Computer.

Dial an octothorpe sign "#", and terminate by dialing the requested wake-up time from the Attendant Console

Manual Line Service

Manual Line or Private Line Services

AWU does not support these features; an AWU call cannot be programmed against a manual line or private line DN.

Multiple Appearance DN

All Multiple Appearance DNs are rung, including both primary and secondary DNs. Programming the wake up request using the Wake Up key applies only to telephones with the primary DN on key 0, and the Wake Up indicator operates as described only on the telephone that is currently programming the wake up request.

In addition, if two or more Multiple Appearance Primary DN telephones program a wake up request at the same time, the last telephone to finish overrides. In other words, all telephones with the same primary DN get the same request time of the last telephone to program a request. If the last telephone cancels the request, all requests are canceled.

When the wake up programming sequence is finished, all Wake Up indicators on Multiple Appearance Prime DNs are updated unless a telephone is in the middle of Wake Up programming.

If the AWU Recall option is chosen, the recall is presented to any idle Attendant Console in the same Console Presentation Group (CPG) equipped with the AWU key.

Night Service

Unanswered AWU calls going through Attendant Recall are discarded if the Attendant Console is in the Night Service mode. AWU may still be programmed when the Attendant Console is in Night Service.

Pretranslation

When the Pretranslation feature is equipped with AWU, the actual DN, not the pretranslation DN, should be used when programming the AWU call request.

Room Status

Room Status and Automatic Wake Up both use the Background Terminal (BGD). If the WAKE option is selected for the check-in/check-out operation, the wake-up call for that room is canceled after a check-in or check-out operation.

When a guest checks in or out, the room status changes. If an AWU request is still active, it is canceled if it is included as part of the Check In/Out option.

Feature packaging

Automatic Wake Up (AWU) package 102 requires:

- Recorded Announcement (RAN) package 7
- Controlled Class of Service (CCOS) package 81
- Background Terminal Facility (BGD) package 99

Guest Entry of Auto Wake Up is included as part of Automatic Wake Up (AWU) package 102.

Multi-Language Wake Up (MLWU) package 206 requires Automatic Wake Up (AWU) package 102.

Multiple Wake Up FFCs require Flexible Feature Codes (FFC) package number 139.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Define the RANF route.

2 LD 16 – Define the RAN1 route.

3 LD 16 – Define the RAN2 route.

4 LD 14 – Define the trunk for RANF.

5 LD 14 – Define the trunk for RAN1.

6 LD 14 – Define the trunk for RAN2.

7 LD 15 – Enable Automatic Wake Up in Customer Data Block.

8 LD 10 – Set language and CCOS for analog (500/2500 type) telephones (on a per TN basis).

9 LD 11 – Set language and CCOS for Meridian 1 proprietary telephones (on a per TN basis).

10 LD 12 – Allow access to AWU from Attendant Consoles.

LD 16 – Define the RANF route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	AWR	AWU RAN route.
RTYP	AUD	Audichron recorder.

- GRD	PLAY IDLE	Ground Start Arrangement where: PLAY = RAN machine sends a ground signal when playing. IDLE = RAN machine sends a ground signal when idle. If the United Kingdom (UK) package 190 is equipped the default response is PLAY, if this package is not equipped the default response is IDLE.
ACOD	xxxx	Trunk route access code. Must be different from RANF ACOD.

LD 16 – Define the RAN1 route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number. Must be different from RANF route number. For Large Systems For Small Systems and Succession 1000 systems
TKTP	AWR	AWU RAN route.
RTYP	AUD	Audichron recorder.

- GRD	PLAY IDLE	Ground Start Arrangement where: PLAY = RAN machine sends a ground signal when playing. IDLE = RAN machine sends a ground signal when idle. If the United Kingdom (UK) package 190 is equipped the default response is PLAY, if this package is not equipped the default response is IDLE.
ACOD	xxxx	Trunk route access code. Must be different from RANF and RAN1 ACODs.

LD 16 – Define the RAN2 route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number. Must be different from RANF and RAN1. For Large Systems For Small Systems and Succession 1000 systems
TKTP	AWR	AWU RAN route.
RTYP	AUD	Audichron recorder.
- GRD	PLAY IDLE	Ground Start Arrangement where: PLAY = RAN machine sends a ground signal when playing. IDLE = RAN machine sends a ground signal when idle. If the United Kingdom (UK) package 190 is equipped the default response is PLAY, if this package is not equipped the default response is IDLE.
ACOD	xxxx	Trunk route access code.

LD 14 – Define the trunk for RANF.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AWR	AWU RAN trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems

LD 14 – Define the trunk for RAN1.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AWR	AWU RAN trunk.
TN	 l s c u c u	Terminal Number. Must be a different TN from RANF. For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number. Must be different from RANF. For Large Systems For Small Systems and Succession 1000 systems

LD 14 – Define the trunk for RAN2.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AWR	AWU RAN trunk.
TN	 l s c u c u	Terminal Number. Must be a different TN from RANF and RAN1. For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number. Must be different from RANF and RAN1. For Large Systems For Small Systems and Succession 1000 systems

LD 15 – Enable Automatic Wake Up in Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	AWU	Automatic Wake Up options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- AWU	YES	Activate AWU for a customer.
- ATRC	(NO) YES	(Deny) allow attendant recall.
CONF	0-159	Conference loop number.
- RANF	0-511	Music RAN route number.
- RAN1	0-511	Primary AWR route number.
- RAN2	0-511 <CR>	Secondary AWR route number.

- LA11	X 0-511	Language 1, RAN route 1. X = remove language RAN route definition.
- LA12	0-511	Language 1, AWR route 2.
- LA21	0-511	Language 2, AWR route 1.
- LA22	0-511	Language 2, AWR route 2.
- LA31	0-511	Language 3, AWR route 1.
- LA32	0-511	Language 3, AWR route 2.
- LA41	0-511	Language 4, AWR route 1.
- LA42	0-511	Language 4, AWR route 2.
- LA51	0-511	Language 5, AWR route 1.
- LA52	0-511	Language 5, AWR route 2.
- R2BN	hhmm	RAN2 start time.
- R2ED	hhmm	RAN2 end time.
- NRWU	2-(5)	Number of rings for a wake up call
- TAWU	1-(3)	Number of wake up tries for an unanswered AWU call

Note 1: AWR route number ranges from 0-511 apply to Large Systems only. Range is 0-127 for all other options. Enter "X" to remove a route.

LD 10 – Set language and CCOS for analog (500/2500 type) telephones (on a per TN basis).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500 2500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LANG	(0)-5	Language number. To remove entry, precede with X.
CLS	CCSA	Controlled Class of Service allowed.

LD 11 – Set language and CCOS for Meridian 1 proprietary telephones (on a per TN basis).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	CCSA	Allow Controlled Class of Service.
LANG	(0)-5	Language number. To remove entry, precede with X.
KEY	xx WUK	Assign a wake up key on a telephone. Must be a key/lamp pair.
Note: To assign a language on a per DN basis, use a Background Terminal.		

LD 12 – Allow access to AWU from Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	aaa	Console type, where: aaa = 1250, 2250.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx AWU	Add an AWU key.

Automatic Wake Up Diagnostic:

To check the availability of the delivery of AWR messages, the technician dials the Access Code (ACOD) from a maintenance set only. A maintenance set is equipped with a MTA Class of Service. Trunks can also be diagnosed in LD 36 by entering the command AWR C R (Test Automatic Wake Up devices associated with Customer (C) and Route (R).

Feature operation

From a telephone with a Wake Up key

To program a wake up call from an idle telephone, follow these steps:

1 Press **Wake Up**.
The indicator flashes.

2 Dial the wake up request time, in 24-hour format (7:30 a.m. as 730, 7:30 p.m. as 1930).
Telephones with display show a dash followed by the time. If no time is set, a single dash is shown.The indicator keeps flashing.

3 Press **Wake Up**.
The indicator goes on steady.

Press the **Release** (Rls) or **PDN** key while programming a wake up request to abort the wake up request. Any previously defined wake up time will remain.

Display telephones If the time interval chosen for the wake up call is full, the system searches for and displays the next available time. If the system cannot find another time, the display shows four dashes (– – – –), and the Wake Up indicator remains flashing. If the system finds another time, the guest has three options:

- To accept the new wake up time, press **Wake Up**.

- To reject the new wake up time and enter another one, dial the new wake up time and press **Wake Up** to validate the new time.
- To abort the wake up time, press **Rls** or the **Prime DN** key (PDN).

To cancel a wake up request, follow these steps:

1 Press **Wake Up**.
The indicator flashes.

2 Dial the octothorpe (#).

3 Press **Wake Up**.
The indicator goes off.

To check a wake up request on a telephone with display, follow these steps:

1 Press **Wake Up**.
The indicator flashes and the current wake up time appears on the display. If no wake up time is programmed, the display shows a dash (–).

2 Press **Wake Up**.
The indicator lights if a wake up time is set.

Note: In each scenario, the Wake Up indicator lights and the display clears, except when the wake up time is aborted and no wake up time was programmed before the abort. In this case, the Wake Up indicator stays off. If a time was programmed before aborting, the previous wake up time is restored, and the indicator is on.

From an Attendant Console

To program a wake up call from an Attendant Console, follow these steps:

1 Press **A. Wake Up**.
The A. Wake Up, ICI, lpk, and S indicators light.

Note: If the displayed number is not the number requiring the wake up call, dial the proper number.

2 Press the **octothorpe (#)**.
If the A. Wake Up indicator remains on steadily, the dialed number is valid. If it flashes, the number is invalid.

3 Dial the requested wake up time using a 24-hour format. Press **A. Wake Up** again.
If the A. Wake Up indicator remains on without flashing, the requested wake up time is acceptable; if it flashes, the time is not acceptable. Enter the new time; if it is acceptable, the indicator goes on without flashing.

4 Press **Rls** to end the procedure.

To cancel a wake up call from an Attendant Console, follow these steps:

1 Press **A. Wake Up**.
The A. Wake Up indicator lights.

Note: If the displayed number is not the number requiring cancellation of the wake up call, dial the proper number.

2 Press the **octothorpe (#)**, then press **A. Wake Up** again.
The A. Wake Up indicator goes off and the wake up request is canceled.

Note: If the indicator flashes quickly, no wake up call was found for the dialed number. Press **A. Wake Up** again.

3 Press **Rls** to end the procedure.

If a guest has not responded after three wake up call attempts, you'll hear a continuous buzz. The indicator will flash quickly. The extension number of the room that has failed to respond will be displayed. Follow these steps:

1 Press **A. Wake Up** to cancel the notification.

2 Press **Rls** to end the procedure.

To Use Multiple Wake Up FFCs

Activate single

The user must dial the Multiple Wake Up Activate (MWUA) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

MWUA H1 H2 M1 M2

Activate repeat (daily)

The user must dial the Multiple Wake Up Repeat Activate (MWRA) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

MWRA H1 H2 M1 M2

Deactivate single

The user must dial the Multiple Wake Up Deactivate (MWUD) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

MWUD H1 H2 M1 M2

Deactivate all

The user must dial the Multiple Wake Up Deactivate (MWUD) FFC:

MWUD H1 H2 M1 M2

Verify

The user must dial the Automatic Wake Up Verify (AWUV) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

AWUV H1 H2 M1 M2

Automatic Wake Up FFC Delimiter

Contents

This section contains information on the following topics:

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Feature description

The Automatic Wake Up Flexible Feature Code Delimiter modifies the user programming interface of the Automatic Wake Up feature, including variations such as Multiple and Repeat Multiple Automatic Wake Up. This modification provides two options for the user: optional delimiter at the end of time entry and optional standard time entry. These options are only applicable to proprietary and analog 2500 sets.

The optional delimiter at the end of time entry during the activation, deactivation or verification of Automatic Wake Up is an octothorpe (#).

The standard time entry allows a customer to enter standard time when activating Multiple Automatic Wake Up. When activated, a customer can eliminate the leading zero when entering a time. For example, the time seven am can be entered as 700 rather than 0700. The time can still be entered with four digits even if the standard time entry option is selected by the customer.

When activated, this feature provides the user with a response from the system. The response is silence or confirmation by means of a tone or a recorded announcement.

Operating parameters

The feature is applicable to all systems.

If the user enables the delimiter option without enabling the standard time entry option, all four digits (H1H2M1M2) and an octothorpe (#) must be entered for a valid entry.

An octothorpe (#) is the only delimiter accepted to indicate the end of time entry. This delimiter is not programmable.

Feature interactions

Background Terminal

When changes to the wake up timer are initiated by the Background Terminal or user, the wake up time previously entered last is overridden. An octothorpe (#) is not required when entering the Wake up time from a background terminal.

Call Detail Recording

No Call Detail Recording report is generated for Automatic Wake Up calls.

Directory Numbers - Multiple Appearance

For Multiple Appearance Directory Numbers, wake up information is stored, deleted and queried from a DN's first primary appearance terminal number.

Directory Number - Prime
Release Key

Pressing the Prime Directory Number or Release key, when programming a Wake up request, cancels the programming sequence. If an invalid timer is entered, the user hears an error tone. If another feature key is pressed during programming, it is ignored by the system.

Room Status

When a guest has either checked in or out, the room status changes. If an AWU request is still active, it is canceled if it is included as part of the Check In/Out option.

Feature packaging

Automatic Wake Up FFC Delimiter requires Flexible Feature Codes (FFC) package 139. The following packages are also required:

- Recorded Announcement (RAN) package 7
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Automatic Wake Up (AWU) package 102

Flexible Tone and Cadences (FTC) package 125 is required if a special error tone rather than overflow is desired for Automatic Wake Up. FTC and Message Intercept (MINT) package 163 is required if a recorded announcement is desired as confirmation from the system after wake up timer has been entered. International Supplementary Features (SUPP) package 131 is required if values other than the default are desired for the inter-digit timer.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Automatic Wake Up in the Customer Data Block.

2 LD 57 – Configure Flexible Feature Codes for Automatic Call Wake Up.

3 LD 56 – Set Automatic Wake Up special error tone and configuration tone.

LD 15 – Enable Automatic Wake Up in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	AWU	Change Automatic Wake Up options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- AWU	YES	Enable Automatic Wake Up data.
...		
WUD	YES	Wake Up Delimiter. IF WUD = YES then time entry is valid only if user enters octothorpe (#) at end of time digits.
STE	YES	Standard Time Entry prompted only if WUD = YES. This prompt permits three or four digit time entry.

LD 57 – Configure Flexible Feature Codes for Automatic Call Wake Up.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
...		
CODE	AWUA	Auto Wake Up activation code.
- AWUA	xxxx	Auto Wake Up activation code for proprietary and Analog (500/2500 type) telephones. AWUA is prompted until <CR> is entered.

...		
CODE	AWUD	Auto Wake Up deactivation code.
- AWUD	xxxx	Auto Wake Up deactivation code for proprietary and Analog (500/2500 type) telephones. AWUD is prompted until <CR> is entered.
CODE	AWUV	Auto Wake Up verification code
- AWUV	xxxx	Auto Wake Up verification code for proprietary and Analog (500/2500 type) telephones. AWUV is prompted until <CR> is entered.
CODE	MWUA	Multiple Wake Up activation.
- MWUA	xxxx	Multiple Wake Up activation code for Analog (500/2500 type) telephones. MWUA is prompted until <CR> is entered.
CODE	MWRA	Repeat Multiple Wake Up activation.
- MWRA	xxxx	Repeat Multiple Wake Up activation code Analog (500/2500 type) telephones. MWRA is prompted until <CR> is entered.
CODE	MWUD	Multiple Wake Up deactivation.
- MWUD	xxxx	Multiple Wake Up deactivation code Analog (500/2500 type) telephones. MWUD is prompted until <CR> is entered.

LD 56 – Set Automatic Wake Up special error tone and configuration tone.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	Flexible Tones and Cadences (FTC) Table Number. To associate a FTC table with trunk route, enter the table number in response to the TTBL prompt in LD 16.

...		
HCCT	YES	Hardware Controlled Cadences and Tone modification of the hardware controlled cadence tone definitions allowed.
...		
- FFCT		Flexible Tone and Cadence.
- - XTON	xxx	Flexible Tone and Cadence confirmation tone. China xxx = 211 North America xxx = 004
- - XCAD	xxx	Cadence code for Firmware Cadence Table (FCAD) as entered at Cadence Number (WCAD) prompt. China xxx = 110 North America xxx = 000
- AWUT		Automatic Wake Up.
- - XTON	xxx	Automatic Wake Up special error tone. China xxx = 214 North America xxx = 007
- - XCAD	xxx	Cadence code for Firmware Cadence Table (FCAD) as entered at Cadence Number (WCAD) prompt. China xxx = 100 North American xxx = 017

Feature operation

The following feature operations occur if the WUD prompt (Wake Up Delimiter) and STE prompt (Standard Time Entry) are set to YES in LD 15. If WUD = YES and STE = NO, then the user must dial all four standard time digits and an octothorpe for a valid entry. If WUD = NO then the STE prompt will not appear. In this case, the prior operation exists and the user is not expected to enter the delimiter (#) at the end of time entry. However, all four time digits must be entered for a valid entry. Table 28 shows the Flexible Feature Codes used in the AWU FFC Delimiter feature.

Table 28
Flexible Feature Codes used in AWU FFC Delimiter feature

Feature	Activation Flexible Feature Code	Deactivation Flexible Feature Code	Verification Flexible Feature Code
Automatic Wake Up (AWU)	AWUA	AWUD	AWUV
Multiple Automatic Wake Up (MAWU)	MWUA	MWUD	AWUV
Repeat Multiple Automatic Wake Up	MWRA	MWUD	AWUV

Flexible Feature Code Automatic Wake Up Activation

To activate Automatic Wake Up from an analog 2500 or a Meridian 1 proprietary telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUA FFC" H1M1M2# or H1H2M1M2#. Get response and go on-hook.

To activate Automatic Wake Up from an analog 500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUA FFC" H1H2M1M2. Get response and go on-hook.

Flexible Feature Code Automatic Wake Up Deactivation

To deactivate Automatic Wake Up from an analog (500/2500) or a Meridian1 proprietary telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUD FFC". Get response and go on-hook.

Flexible Feature Code Multiple Automatic Wake Up Activation

To activate Multiple Automatic Wake Up from an analog 2500 set:

1 Go off-hook. Listen for dial tone.

2 Dial “MWUA FFC” H1M1M2#. Get response and go on-hook.

3 Repeat for up to four wake up times maximum per day.

To activate Multiple Automatic Wake Up time from an Analog 500 set:

1 Go off-hook. Listen for dial tone.

2 Dial “MWAU FFC” H1H2M1M2. Get response and go on-hook.

Flexible Feature Code Multiple Automatic Wake Up Deactivation

To deactivate single wake up time from an analog 2500 set:

1 Go off-hook. Listen for dial tone.

2 Dial “MWUD FFC” H1M1M2# or H1H2M1M2#. Get response and go on-hook.

3 Repeat for other wake up times as necessary.

To deactivate a single wake up time from an analog 500 set:

1 Go off-hook. Listen for dial tone.

2 Dial “MWUD FFC” H1H2M1M2. Get response and go on-hook.

3 Repeat for other wake up times as necessary.

To deactivate all wake up times from an analog 2500 telephone:

1 Go off-hook. Listen for dial tone.

2 Dial “MWUD FFC” #. Get response and go on-hook.

To deactivate all wake up times from an analog 500 telephone:

1 Go off-hook. Listen for dial tone.

2 Dial “MWUD FFC” and go on-hook.

Flexible Feature Code Automatic/Multiple Automatic Wake Up Verification

To verify Automatic/Multiple Automatic Wake Up from an analog 2500 set:

1 Go off-hook. Listen for dial tone.

2 Dial “AWUV FFC” H1M1M2# or H1H2M1M2. Get response and go on-hook.

3 Repeat for other wake up times as necessary.

To verify Automatic/Multiple Automatic Wake Up from an analog 500 set:

1 Go off-hook. Listen for dial tone.

2 Dial “AWUV FFC” H1H2M1M2. Get response and go on-hook.

Auxiliary Processor Link

Contents

This section contains information on the following topics:

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Operating parameters 531
Feature interactions 532
Feature packaging 532
Feature implementation 532
Feature operation 532

Feature description

The Auxiliary Processor Link (APL) is a full-duplex asynchronous data link capable of accommodating up to a 4800 baud rate. It is connected to the system through a Serial Data Interface (SDI) port.

This feature is currently used in conjunction with the Integrated Messaging System package and the Automatic Call Distribution (ACD) Dialed Number Identification Service (DNIS) package.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

DNIS Length Flexibility

Expanded DNIS (more than four DNIS digits) is not supported on the APL.

Feature packaging

Auxiliary Processor Link (APL) package 109 has no feature package dependencies.

Feature implementation

To implement this feature, refer to *Automatic Call Distribution: Description* (553-3001-351).

Feature operation

No specific operating procedures are required to use this feature.

Auxiliary Signaling

Contents

This section contains information on the following topics:

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Feature operation 534

Feature description

In some situations, customers require special auxiliary devices such as bells, buzzers, or lights to be connected through the system. These devices are activated through a regular 500/2500 Line Card and its associated data block.

Operating parameters

A C4A ringer, or any other special signaling device that can be activated by a 20 Hz ringing signal, can be equipped through the 500/2500 Line Card.

A maximum of five C4A ringers or equivalent devices can be configured on one Terminal Number. This limit depends on the device's impedance to the 20 Hz ringing.

Feature interactions

Mixed DNs

If the DN associated with the signaling device appears on analog (500/2500 type) or Meridian 1 proprietary telephones, the telephone can answer or connect into an active call.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 10 – Add new 500 telephone data block.

Prompt	Response	Comment
REQ:	NEW	Add new data block.
TYPE:	500	Analog (500/2500 type) telephone
...		
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		

Feature operation

No specific operating procedures are required to use this feature.

B34 Codec Static Loss Plan Downloading

Contents

This section contains information on the following topics:

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Feature packaging 542
Feature implementation 542
Feature operation 544

Feature description

This feature provides software support for Static Loss Plan Downloading to the B34 codec. A codec is a device on an Intelligent Peripheral Equipment (IPE) card which encodes incoming transmission data from analog to digital, and decodes outgoing transmission data from digital to analog.

The B34 codec is a four-channel codec providing 32 programmable loss values in 0.5 dB steps in both the transmit and receive directions. The B34 allows transmission parameters, which can be downloaded to the IPE unit, to be changed by software. Since the loss and level requirements differ from country to country, this allows system compliance to the different transmission plans used in the world markets using a single codec.

The selected coded levels are downloaded to each unit based on the unit's port type classification at initialization, configuration, or enable time. This is referred to as static downloading. These levels will be used for all call connections involving that unit. The B34 Codec Static Loss Plan Downloading feature is used on systems where a single loss setting is sufficient for all types of call connections.

Some markets, however, require adjustments on the loss setting depending on the call connection. This is referred to as Dynamic Pad Switching, or Dynamic Loss Switching, and is addressed by the Dynamic Loss Switching feature, which is described elsewhere in this publication. The B34 Codec Static Loss Plan Downloading feature provides the basis for the Dynamic Loss Switching feature; if Dynamic Loss Switching is enabled for a system, Static Loss Plan Downloading is suppressed on that same system.

The transmission plan for each country follows the European Telecommunications Standards Institute (ETSI) standard of loss values (referred to as "new values"), or existing values (loss values currently provided by existing cards in ETSI countries). New IPE cards must be capable of accepting these existing values for use in existing systems, so as to maintain port-to-port loss integrity.

Typically, existing (pre-Phase 8B) systems do not require flexible B34 equipped IPE cards unless their loss plans change; these systems use the existing loss plans. New systems installed with Phase 8B software contain only flexible B34 equipped IPE cards, and can use either existing loss plans or the ETSI loss plans. Systems equipped with both flexible B34 equipped IPE cards and non-B34 equipped IPE cards require type approval to be secured under existing loss plan values.

The Static Loss Plan Download feature allows the selection of a loss plan table which is either compliant with the old or the new loss plan for various countries. The feature is supported on international IPE analog trunk cards (XCOT, XFCOT, XDID, XCO/XDID, XFEM, or any trunk configured with XTRK type of XCOT, XDID, or XFEM) with the right firmware support. In special situations and with the right authorization, a customized table may be defined.

When selecting a loss plan table, it is important to verify whether the existing or ESTI mode is to be exhibited by the system (the “Feature implementation” section explains how to install a loss plan table using LD 97). A Service Change interface allows an existing or ETSI table to be selected by specifying a loss plan table number. If the loss plan needs to be upgraded in the field or if a newly defined loss plan has to be installed, a service change may be performed by an authorized craftsperson to enter a table of customized loss plan values for each port type, or to customize a pre-defined table by changing the table values. The table can then be downloaded upon any of the following conditions:

- at system initialization for all units
- when a trunk or line card, or trunk or line unit is enabled
- when the XPEC is enabled
- when the IPE shelf is enabled
- when a configured card is reset
- after a trunk unit has undergone a “NEW”, “CHANGE”, or “MOVE” operation using LD 14 or LD 10

There is no mechanism to indicate whether or not an IPE card is equipped with the B34 codec. Loss plan messages are downloaded to all IPE cards in hybrid systems, whether or not they are equipped with the B34 codec. Typically, there are three vintages of firmware used in the field:

- non-B34 codec equipped cards
- hardcoded B34 equipped cards
- flexible B34 equipped cards

There are two versions of the flexible B34 equipped cards, a flexible 7C software compatible B34 equipped card and a flexible 8B software compatible B34 equipped card. The hardcoded and 7C software compatible versions of the B34 equipped cards have country-unique loss value defaults. The flexible 8B software compatible B34 equipped card have universal B34 default loss values, which do not meet any country-specific requirements.

The flexible 7C software compatible B34 equipped card and the flexible 8B software compatible B34 equipped card both recognize the new B34 (type 12) messages, as well as the old static pad switching (type 5) messages; the type 12 messages take precedence. The hardcoded B34 cards only recognize the type 5 messages.

The flexible 7C B34 equipped cards are forward compatible with the new software; the hardcoded B34 cards are not. The flexible 8B B34 equipped cards are not backwards compatible to systems running older versions of software.

Operating parameters

A system must be configured with one or more IPE cards equipped with a B34 codec and firmware supporting software downloading. It is the responsibility of the installer to verify that the IPE cards used are compliant with the download messages used by this feature.

XFALC (flexible analog line card) is compatible with the download messages supporting Static Loss Plan Download.

Since the flexible 8B B34 equipped cards are not backwards compatible to systems running older versions of software, the following upgrade strategy should be followed:

- Systems running software Phase 7C or earlier, and upgrading to Phase 8B software, do not require the new flexible B34 IPE cards if the transmission plan remains the same. These systems may be equipped with a mix of hardcoded B34 IPE cards and new flexible B34 IPE cards; if changing to the new ETSI loss plan, all hardcoded B34 IPE cards must be retrofitted with the new flexible B34 IPE cards.
- Systems changing to a new ETSI loss plan must use the new flexible B34 IPE cards as well as Phase 8B static parameter download software; a hardware retrofit and a software upgrade are also necessary.
- Newly installed systems will use the new flexible B34 IPE cards.

New flexible B34 equipped XFALC (flexible analog line cards) support Static Loss Plan Downloading using B34 messages. New flexible B34 equipped XFALCs installed in a Phase 7C software environment do not receive download messages, but use the firmware-defined default.

A distinction must be made between long and short lines on ALC units, and to download loss plan values based on this setting.

Feature interactions

Alternative Loss Plan

The alternative loss plan tables must be enlarged as the default table is enlarged.

B34 Dynamic Loss Switching

B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching. Both features share the same definition of port types and use the same base-level table.

When B34 Dynamic Loss Switching is enabled, the Static Download messages to the analog trunk cards are suppressed. Static download to analog line cards continues.

B34 codec static loss download. Since the B34 Dynamic Loss Switching is dependent on B34 Codec Static Loss Download, B34 Codec Static Loss Download must be enabled when B34 Dynamic Loss Switching is enabled. The port types defined for B34 Dynamic Loss Switching are a subset of the port types defined for B34 Codec Static Loss Download.

Also, the base level table used by B34 Codec Static Loss Download is also used by B34 Dynamic Loss Switching. Since B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching, B34 Codec Static Loss Download is enabled when B34 Dynamic Loss Switching is enabled. When B34 Dynamic Loss Switching is enabled, the following operations concerning trunk cards are suppressed:

- During initialization, B34 Codec Static Loss Downloading to trunk cards is suppressed, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- In LDs 32 and 36, B34 Codec Static Loss Download is suppressed on enabling the trunk card or unit, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- When reseating the cards, B34 Codec Static Loss Download is suppressed. Downloading continues to analog line cards.

- In LD 14, B34 Codec Static Loss Download is suppressed.
- In LD 10, B34 Codec Static Loss Download is suppressed.

When B34 Dynamic Loss Switching is disabled, all B34 Codec Static Loss Downloads to trunks are suppressed. This introduces the danger of having some cards in the system which are not set with the proper loss levels, since the system has been changed from a dynamic mode to a static mode without activating the download of the static messages. To highlight this change, a SCH5842 error message is generated, indicating to the craftsperson that B34 Dynamic Loss Switching is disabled, and that B34 Static Loss Downloading is now in effect and that a download should be activated by system initialization or SYSLOAD.

When B34 Dynamic Loss Switching is enabled, all B34 Codec Static Loss Download audit messages to trunk cards are suppressed.

Conference

When a conference connection is established, no pads are switched in on the trunk side; any extra loss that is required is provided by the conference circuit based on an algorithm which takes into account the number of lines and trunks.

Digital Trunk Interface (DTI) Pad Switching

Pad switching for DTI applications is done dynamically, based on the far end's port type. On the DTI side, a loss value is switched on the receive and transmit side, depending on the far end's port type. If the far end is analog, a pad is switched in or out; if the far end is digital, a zero loss is switched in, so that the relative loss is taken care of only on one side. Connection between DTI/PRI and XDID, XFCOT, and XFEM trunks is not supported, since DTI pad switching does not take care of these trunk types.

DTI2 Pad Switching

Pad switching for DTI2 applications is done dynamically, based on the far end's port type. On the DTI side, a loss value is switched on the receive and transmit side, depending on the far end's port type. The far end side is handled by the normal operation for the trunk type. That is, if the far end is DTI, it is handled according to the DTI pad switching. If it is otherwise, it is handled by the configured matrices. No messages are sent for XDID, XFCOT, and XFEM trunk types. For XUT and XEM trunk types, the loss equivalent to pad out is switched in. For XDID, XFCOT, and XFEM trunks, the base level (static) value is switched in when connected to the DTI2 trunk types.

GEC Plessy hardware

No losses are sent to XCOT, XDID, and XFEM trunk cards when these cards are connected to GEC hardware, since there is no dynamic switching done for them. On the GEC hardware side of such a connection, the pads are switched in according to the type of trunk (near end) as opposed to what it is connected (far end); therefore, the loss is switched in regardless of whether the connection is to XCOT, XDID, and XFEM trunk cards or other types of cards.

Intelligent Peripheral Equipment Completion

Whenever a TIE/LDR trunk is configured on an XIDID card, for Static Loss Plan Download (SLPD)/Dynamic Loss Switching (DLS), loss/level is downloaded/switched to an XDID card with the type 12 message. Depending on the Class of Service configured, Non-Transmission Compensated (NTC), Transmission Compensated (TRC), or Via Net Loss (VNL), the TIE unit will be mapped to the following B34 port types: B34 T2WN, B34 T2WT, or B34 T2WV.

ISDN Basic Rate Interface

It is possible to switch in loss on the ISDN BRI side, based on port types.

MFE/MFC Pads

The Alternative Loss Plan feature allows trunks to be configured so as to have pads switched in when an MFS sender/receiver is equipped. For such a configuration, the following occurs for B34 port types:

- Pads are switched in for outgoing calls (the trunk is the originator).
- Pads are switched in, if in the dialing state, for incoming calls (the trunk is the terminator).

Feature packaging

B34 Codec Static Loss Plan Downloading requires Intelligent Peripheral Equipment (XPE) package 203.

Feature implementation

The loss level tables are configured in LD 97. The craftsperson must have an authorized password to configure the loss tables, but printing of the tables can be performed without the password.

LD 97 – Configure a loss plan table.

Prompt	Response	Description
TYPE	LOSP	The type branch for the system loss plan table.
STYP	(PRED) CSTM DISL	The type of B34 static loss plan table to be used to download B34 programmable loss codes. Enter PRED if a numbered pre-defined static loss plan is to be used. Enter CSTM to customize an existing static loss plan table by modifying one or more existing entries, or to create a new table by entering new values to all entries. Enter DISL to disable static loss plan downloading.

PWD2	xxxx	Enter the level 2 administrator password. Note that this is prompted only when STYP=DISL or STYP=CSTM. If STYP=DISL, and the proper password is entered, then the next prompt is REQ. If STYP=CSTM, and the proper password is entered, then the next prompts are the PORTTYPES (for example, COTS, COTL). If the password entered is incorrect, an existing error message, SCH523, SCH525, SCH526 will be issued and PWD2 will be re-prompted.
TNUM	1-25	Prompted only if PRED was entered in response to the STYP prompt above. Enter the number for the required pre-defined static loss plan.
COTS	Rx Tx	Prompted only if the response to the STYP prompt above was CSTM. COT short line. Enter the coded input/output relative levels in the receive (Rx) direction and in the transmit (Tx) direction, for this port type. The input range of Rx and Tx for port types associated with trunks is 8-39 and 0-31 respectively; the input range of Rx and Tx for port types associated with analog lines is 0-31 and 8-39 respectively.
COTL	Rx Tx	COT long line. The same definition applies as for COTS.
DIDS	Rx Tx	DID/DOD short line. The same definition applies as for COTS.
DIDL	Rx Tx	DID/DOD long line. The same definition applies as for COTS.
T2WT	Rx Tx	TIE, 2 wire, Class of Service TRC. The same definition applies as for COTS.
T2WN	Rx Tx	TIE, 2 wire, Class of Service NTC. The same definition applies as for COTS.
T2WV	Rx Tx	TIE, 2 wire, Class of Service VNL. The same definition applies as for COTS.
T4WT	Rx Tx	TIE, 4 wire, Class of Service TRC. The same definition applies as for COTS.
T4WN	Rx Tx	TIE, 4 wire, Class of Service TRC. The same definition applies as for COTS.

T4WV	Rx Tx	TIE, 4 wire, Class of Service VNL. The same definition applies as for COTS.
PAGT	Tx	TIE, E&M 2 paging trunk. The same definition for Tx applies as for COTS. Note that there is no loss value associated with this trunk type in the receive (Rx) direction.
RANR	Rx	Recorded Announcement Route. The same definition for Rx applies as for COTS. Note that there is no loss value associated with this trunk type in the transmit (Tx) direction. Enter the Coded Receive (A/D) Input/Output level, where Rx = 8-39.
ALUS	Rx Tx	ALC unit short line (SHL) Class of Service. Enter the coded input/output relative levels in the receive (Rx) direction and in the transmit (Tx) direction, for this port type. The input range of Rx and Tx for port types associated with analog lines is 0-31 and 8-39 respectively.
ALUL	Rx Tx	ALC unit long line (LOL) Class of Service. Enter the coded input/output relative levels in the receive (Rx) direction and in the transmit (Tx) direction, for this port type. The input range of Rx and Tx for port types associated with analog lines is 0-31 and 8-39 respectively.

Feature operation

No specific operating procedures are required to use this feature.

B34 Dynamic Loss Switching

Contents

This section contains information on the following topics:

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Feature description

A codec is a device on an Intelligent Peripheral Equipment (IPE) card which encodes incoming transmission data from analog to digital, and decodes outgoing transmission data from digital to analog. The B34 codec is a four-channel codec providing 32 programmable loss values in 0.5 dB steps in both the transmit and receive directions. The B34 allows transmission parameters, which have been downloaded to the IPE unit, to be changed by software. Since the loss and level requirements differ from country to country, this allows system compliance to the different transmission plans used in the world markets using a single codec. The selected coded levels are downloaded to each unit based on the unit's port type classification. This is referred to as static downloading. These levels will be used for all call connections involving that unit. The B34 Codec Static Loss Plan Downloading feature is, therefore, used on systems where a single loss setting

is sufficient for all types of call connections (this feature is described elsewhere in this publication).

The Dynamic Loss Switching feature provides loss switching on international IPE analog trunks cards (XCOT, XFCOT, XDID, XCO/XDID, XFEM, or any trunk configured with XTRK type of XCOT, XDID, or XFEM).

Typically, there are different vintages of firmware in the field:

- hard-coded B34 firmware, which is hardcoded with country-specific defaults, ignores B34 type 12 messages, and accepts (where applicable) Short Line/Long Line configuration type 5 messages
- flexible B34 firmware with country-specific defaults, which is firmware that is coded with country-specific defaults, accepts (where applicable) Short Line/Long Line configuration type 5 messages, and accepts B34 type 12 messages which override any accepted Short Line/Long Line configuration type 5 messages
- flexible B34 firmware with universal defaults, which is firmware that is coded with a universal B34 loss value default, may or may not ignore Short Line/Long Line configuration type 5 messages, and accepts B34 type 12 messages which override any accepted Short Line/Long Line configuration type 5 messages

To obtain the full functionality of B34 Dynamic Loss Switching, only the two flexible vintages of firmware can be used.

Every time a new connection is established, the following process is followed to determine if and how to adjust the loss involved in the connection:

- the port type of the originator and terminator is determined, based on the configurations of the originator and terminator, respectively
- this port type is used as a row index (originator) and column index (terminator) into a connection matrix, to determine the following:
  - whether to switch the pad in or out for the originator receive direction
  - whether to switch the pad in or out for the originator transmit direction
  - whether to switch the pad in or out for the terminator receive direction

— whether to switch the pad in or out for the terminator transmit direction

- a message conveying this information is then sent to the originator and terminator, if they are affected port types.

The B34 Dynamic Loss Switching feature, configured on a system basis, introduces flexibility in the loss values to be switched. Where previously the loss values were hardcoded on the analog trunk cards, they are now software-configurable on a per-system basis. The loss switching is still controlled by a connection matrix defined for specific markets. This matrix cannot be changed. The loss levels to be used are configured in a base-level table and alternative-level table in LD 97. The base level table is the same as the one implemented and used by the B34 Static Loss Plan Downloading feature; the alternative level table is a parallel table configured for the B34 Dynamic Loss Switching feature.

These new port types reside on the international IPE cards with flexible B34 firmware and the B34 codec. They have to be distinguished from existing port types because of the different manner in which they are informed of the base level/alternative level information.

Operating parameters

A system must be configured with one or more IPE card equipped with a B34 codec and firmware supporting software downloading. It is the responsibility of the installer to verify that the IPE cards used are compliant with the download messages used by this feature.

The B34 Codec Static Loss Plan Downloading feature must be equipped, since the B34 Dynamic Loss Switching feature uses its base level table.

Since the flexible 8B B34 equipped cards are not backwards compatible to systems running older versions of software, the following upgrade strategy should be followed:

- Systems running software Phase 7C or earlier, and upgrading to Phase 8B software, do not require the new flexible B34 IPE cards if the transmission plan remains the same. These systems may be equipped with a mix of hardcoded B34 IPE cards and new flexible B34 IPE cards; if changing to the new European Telecommunications Standards Institute (ETSI) loss plan, all hardcoded B34 IPE cards must be retrofitted with the new flexible B34 IPE cards.
- Systems changing to a new ETSI loss plan must use the new flexible B34 IPE cards as well as Phase 8B static parameter download software; a hardware retrofit and a software upgrade are also necessary.
- Newly installed systems will use the new flexible B34 IPE cards.

XFALC (Flexible Analog Line Card) is compatible with the download messages supporting Static Loss Plan Downloading. XFALC is not supported in Dynamic Loss Switching.

Connection matrixes are supported for Australia, New Zealand, and Italy. No other countries are supported with this feature.

New flexible B34 equipped XFALC (flexible analog line cards) support Static Loss Plan Downloading using B34 messages. New flexible B34 equipped XFALCs installed in a Phase 7C software environment do not receive download messages, but use the firmware-defined default.

A distinction must be made between long and short lines on Analog Line Cards (ALC), and to download loss plan values based on this setting.

Feature interactions

Alternative Loss Plan

The alternative loss plan tables must be enlarged as the default table is enlarged.

B34 Codec Static Loss Download

B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching. Both features share the same definition of port types and use the same base-level table.

When B34 Dynamic Loss Switching is enabled, the Static Download messages to the analog trunk cards are suppressed. Static download to analog line cards continues.

B34 codec static loss download. Since the B34 Dynamic Loss Switching is dependent on B34 Codec Static Loss Download, B34 Codec Static Loss Download must be enabled when B34 Dynamic Loss Switching is enabled. The port types defined for B34 Dynamic Loss Switching are a subset of the port types defined for B34 Codec Static Loss Download.

Also, the base level table used by B34 Codec Static Loss Download is also used by B34 Dynamic Loss Switching. Since B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching, B34 Codec Static Loss Download is enabled when B34 Dynamic Loss Switching is enabled. When B34 Dynamic Loss Switching is enabled, the following operations concerning trunk cards are suppressed:

- During initialization, B34 Codec Static Loss Downloading to trunk cards is suppressed, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- In LDs 32 and 36, B34 Codec Static Loss Download is suppressed on enabling the trunk card or unit, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- When reseating the cards, B34 Codec Static Loss Download is suppressed. Downloading continues to analog line cards.
- In LD 14, B34 Codec Static Loss Download is suppressed.
- In LD 10, B34 Codec Static Loss Download is suppressed.

When B34 Dynamic Loss Switching is disabled, all B34 Codec Static Loss Downloads to trunks are suppressed. This introduces the danger of having some cards in the system which are not set with the proper loss levels, since the system has been changed from a dynamic mode to a static mode without activating the download of the static messages. To highlight this change, a SCH5842 error message is generated, indicating to the craftsperson that B34 Dynamic Loss Switching is disabled, and that B34 Static Loss Downloading is now in effect and that a download should be activated by system initialization or SYSLOAD.

When B34 Dynamic Loss Switching is enabled, all B34 Codec Static Loss Download audit messages to trunk cards are suppressed.

Conference

When a conference connection is established, no pads are switched in on the trunk side; any extra loss that is required is provided by the conference circuit, based on an algorithm which takes into account the number of lines and trunks.

Digital Sets Transmission Parameters

The following static parameters, which do not change on a connection basis, can be changed using LD 17:

- sidetone objective loudness rating
- receive objective loudness rating
- transmit objective loudness rating
- handsfree receive objective loudness rating
- handsfree transmit objective loudness rating
- handsfree receive objective loudness rating

Digital Trunk Interface (DTI) Pad Switching

Pad switching for DTI applications is done dynamically, based on the far end's port type. On the DTI side, a loss value is switched on the receive and transmit side, depending on the far end's port type. If the far end is analog, a pad is switched in or out; if the far end is digital, a zero loss is switched in, so that the loss is taken care of only on one side. Connection between DTI/PRI and XDID, XFCOT, and XFEM trunks is not supported, since DTI pad switching does not take care of these trunk types.

The far end side is handled by the normal operation for the trunk type: that is, if the far end is DTI, it is handled according to the DTI pad switching, if it is otherwise, it is handled by the configured matrices. No messages are sent for XDID, XFCOT, and XFEM trunk types. For XUT and XEM trunk types, the loss equivalent to pad out is switched in. For XDID, XFCOT, and XFEM trunks, the base level (static) value is switched in when connected to the DTI2 trunk types.

Echo Suppression

When the echo suppresser is turned on for XEM and XFEM trunks, the pad is switched to out. For XEM and XFEM trunks with B34 port types, the base loss level for the affected port type is switched in to match the operation of switching out the pad.

GEC Plessy Hardware

No losses are sent to XCOT, XDID, and XFEM trunk cards when these cards are connected to GEC hardware, since there is no dynamic switching done for them. On the GEC hardware side of such a connection, the pads are switched in according to the type of trunk (near end) as opposed to what it is connected (far end); therefore, the loss is switched in regardless of whether the connection is to XCOT, XDID, and XFEM trunk cards or other types of cards.

ISDN Basic Rate Interface

It is possible to switch in loss on the ISDN BRI side, based on port types.

MFE/MFC Pads

The Alternative Loss Plan feature allows trunks to be configured so as have pads switched in when am MFC sender/receiver is equipped. For such a configuration, the following occurs for B34 port types:

- pads are switched in for outgoing calls (the trunk is the originator), or
- pads are switched in, if in the dialing state, for incoming calls (the trunk is the terminator).

Off Premise Extension Pad Switching

Pads can be switched on an Off Premise Extension card depending on the type of connection.

XCOT, XFEM, and XDID Cards

XCOT, XFEM, and XDID cards are the suite of international IPE cards which are configured under the XTRK prompt in LD 14. The cards in this suite include XDID/DOD, XFCOT, XFEM, XDID, and XCOT. When B34 Dynamic Loss Switching is enabled, these cards receive B34 messages. Since certain markets do not desire this functionality, B34 Dynamic Loss Switching should not be enabled.

During lamp audit for active calls on XCOT, XFEM, and XDID cards, a type 5 message for pad switching is sent to these cards, based on their configuration. When B34 Dynamic Loss Switching is enabled, the type 5 message is not sent; instead, a B34 message is sent, based on the last loss switching message sent for that call.

XEM and XUT Cards

XEM and XUT cards are the suite of North American IPE cards which are configured under the XTRK prompt in LD 14. The cards in this suite include XUTJ, XUT Hong Kong, XEM, and XUT. When B34 Dynamic Loss Switching is not enabled, there is no change in the operation of pad switching on these cards. When B34 Dynamic Loss Switching is enabled, the expanded portion of the connection matrix is used to determine the processing on the XEM/XUT side of the call and on the B34 port type side of the call. When a decision is made, it is communicated using a B34 message.

Feature packaging

B34 Dynamic Loss Switching requires the following packages:

- International Supplementary Features (SUPP) package 131
- Limited Access to Overlays (LAPW) package 164
- Intelligent Peripheral Equipment (XPE) package 203

Feature implementation

The base and alternate tables are configured in LD 97. The connection matrix is selected in LD 15.The craftsperson must have an authorized password to configure the loss tables. Printing of the tables can be performed without the password.

Note: The system must be configured with the Limited Access to Overlays (LAPW) package, and the craftsperson must have an authorized password.

LD 97 – Configure a loss plan table.

Prompt	Response	Description
REQ	CHG	Change loss plan table.
TYPE	LOSP	The type branch for the system loss plan table. Enter LOSP.
...		

Feature operation

No specific operating procedures are required to use this feature.

Background Terminal

Hospitality and health care personnel use Background Terminal (BGD) to enter, retrieve, and modify data associated with the following features:

- Automatic Wake Up (AWU)
- Room Status (RMS)
- Message Registration (MR)
- Call Party Name Display (CPND)

BGD helps monitor system operations by providing a visual display of information changes, hard-copy backup, and traffic statistics.

For complete information on this feature, refer to *Hospitality Features: Description and Operation* (553-3001-353).

Boss/Secretary Filtering Enhancement

Contents

This section contains information on the following topics:

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Feature operation 564

Feature description

The Boss/Secretary Filtering Enhancement (BSFE) feature is designed for a boss/secretary environment.

Prior to the introduction of the BSFE feature, a boss could forward incoming calls to secretary/secretaries for screening.

With the BSFE feature, incoming calls are forwarded from the boss to a designated secretary using the Call Forward and Busy Status (BFS) key. A maximum of 16 BFS keys can be configured on the boss set. A corresponding BFS key is configured on each secretary set. The following enhancements are also introduced by this feature:

- Display capabilities: If the Display key is pressed during an incoming filtered call, the calling party's name and number appear on the telephone display.

- Transfer capabilities: If a secretary presses the BFS key once, listens for the boss to pickup and presses the BFS key a second time, the incoming filtered call is transferred back to the boss.
- New Classes of Service: The Boss Secretary Filtering Enhancement Class of Service Allowed (BFEA) or Denied and the Recall to Boss Allowed (RCBA) or Denied (RCBD).
- Key Lamp status: The BSFE feature allows configuration of the LCD indicator for the BFS key. It is possible to configure the same LCD lamp status to
  - — Dark (key lamp is off)
  - — Lit (key lamp is steadily lit)
  - — Wink
  - — Flash (continual flash of light, 60 ipm)

The BSFE feature is configured on the boss set, with a defined BFS key for each secretary that the boss may select to filter the boss' incoming calls. The set will also have a designated key matching the boss key. The BFS key must be a single appearance DN for the boss and the secretary sets. The BFS keys for the boss/secretary sets are configured in pairs and are on the same node.

Operating parameters

Proprietary telephones with display support the BSFE feature. The BSFE feature cannot be configured for analog (500/2500) telephones or Integrated Services Digital Network (ISDN) BRI telephones. The ringing appearance of the DN can be on an analog (500/2500) telephone but not for a private line.

The BSFE feature cannot be activated simultaneously with the following features:

- Call Forward and Busy Status
- Call Forward All Calls
- Remote Call Forward
- Flexible Feature Code Boss Secretarial Filtering

The BSFE feature supports a maximum of 16 secretary sets associated with the boss set.

With the BSFE feature, the BFS key of the boss is generally non-ringing with key lamp indication notification; the secretary set is set up as ringing.

The BSFE feature cannot be activated if the DN of either set is configured as an Automated Call Distribution (ACD) key.

Feature interactions

Hold

If the BSFE feature is active, the secretary answers the incoming boss call by pressing the SCR key or by pressing the BFS key. If the call is answered on the BFS key, pressing the key a second time will automatically put the call on hold and autodial the DN of the boss. If the class of service of the set is Auto Hold Allowed (AHA) and the call is on the BFS key, pressing the SCR key a second time puts the call on hold. If the class of service of the set is Auto Hold Denied (AHD) and the call is on the BFS key, pressing the SCR key again releases the call.

Hotline

Hotline takes precedence over BSFE. Hotline calls to the boss set are not filtered, even if the BSFE feature is active. The hotline calls are directed to the boss set.

Voice Call

If the Voice Call key/lamp is configured as the boss DN on a third party's set, the call is not filtered by the BSFE feature and the call terminates on the boss set.

Voice Mail

If a call is unanswered, whether the BSFE feature is active or deactivated, the voice mail message is sent directly to the voice mail box of the boss.

The BSFE feature takes precedence over the following features:

- **Camp On**
  If the BSFE feature is active on the boss set, the incoming calls are not camped on this DN but are sent directly to the secretary set.
- **Call Waiting**
  If a call comes in while the boss is on a call and the BSFE feature is active, the call is sent directly to the secretary set.
- **Call Forward and Hunt Override**
  If a secretary calls the boss without using the Call Forward and Busy Status (BFS) key, the call goes back to the secretary. If the secretary uses the BFS key when calling the boss, the call goes to the primary DN of the boss.
- **Do Not Disturb**
  If the BSFE feature is active on the boss set, the Do Not Disturb (DND) is overridden and the call is sent directly to the secretary.
- **Hunting**
  If the boss has Hunt configured and the BSFE feature is active, an incoming call is forwarded to the secretary, not sent through the hunt chain. If the secretary set is busy, the call follows the secretary hunt list.
- **Make Set Busy**
  If the BSFE feature and the MSB key is active, the incoming call is sent directly to the secretary; the caller does not receive a busy tone.
- **Private Line**
  Private Line calls are filtered by the secretary if the BSFE feature is active.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the lamp status for the Boss/Secretary Filtering Enhancement feature.

2 LD 15 – Configure Offhook Alarm Security.

3 LD 11 – Configure the Boss/Secretary Filtering Enhancement feature for Meridian proprietary sets.

Note: The technician must be aware of the various configurations allowed for the LCD lamp notification states (dark, lit, wink, and flash) to avoid user confusion. The default lamp status states are shown below.

Boss set	Boss set with BFS deactivated	Boss set with BFS activated
Idle Busy	Dark ▷ Lit ▶	Wink ▶ Flash ▶

LD 15 – Configure the lamp status for the Boss/Secretary Filtering Enhancement feature.

Prompt	Response	Description
REQ:	CHG	Change exisiting data.
TYPE:	FTR	Features and Options data.
CUST	 0-99 0-31	Customer Number For Large Systems For Small Systems and Succession 1000 systems
...	...	
BSFE	YES	YES = Allow Boss/Secretary Filtering Enhancement feature. (NO) = Deny Boss/Secretary Filtering Enhancement feature.

- ACT_IDLE		Boss's Lamp status when BSFE is active and set is idle.
	(WINK)	LCD Lamp flash rate is 60 impulses per minute.
	FLSH	LCD Lamp flash rate is 30 impulses per minute.
	LIT	LCD Lamp is on.
	DARK	LCD Lamp is dark.
- ACT_BUSY		Boss's Lamp status when BSFE is active and set is busy.
	(FLSH)	LCD Lamp flash rate is 30 impulses per minute.
	WINK	LCD Lamp flash rate is 60 impulses per minute.
	LIT	LCD Lamp is on.
	DARK	LCD Lamp is dark.
- DACT_IDLE		Boss's Lamp status when BSFE is disabled and set is idle.
	(DARK)	LCD Lamp is dark.
	WINK	LCD Lamp flash rate is 60 impulses per minute.
	LIT	LCD Lamp is on.
	FLSH	LCD Lamp flash rate is 30 impulses per minute.
- DACT_BUSY		Boss's Lamp status when BSFE is disabled and set is busy.
	(LIT)	LCD Lamp is on.
	WINK	LCD Lamp flash rate is 60 impulses per minute.
	FLSH	LCD Lamp flash rate is 30 impulses per minute.
	DARK	LCD Lamp is dark.
...	...	

LD 15 – Configure Offhook Alarm Security.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	OAS	Off-Hook Alarm Security options.
CUST		Customer Number
	0-99	For Large Systems
	0-31	For Small Systems and Succession 1000 systems
ODN0	xxxx	Offhook Alarm Security for zone 0

LD 11 – Configure the Boss/Secretary Filtering Enhancement feature for Meridian proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of proprietary set
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	x..x	Office Data Administration System Designator
CUST	xx	Customer number, as defined in LD 15
...	...	
CLS	BFEA	BFEA = Allow Boss/Secretary Filtering Enhancement for set. (BFED) = Deny Boss/Secretary Filtering Enhancement for set.
CLS	RCBA	RCBA = Allow Recall to boss on set basis. (RCBD) = Deny Recall to boss on set basis. ***Note:*** This class of service forwards unanswered calls back to the boss after a specified number of rings.
...	...	
KEY	 xx BFS l s c u xx BFS c u	Call Forward and Busy Status (BFS) key. xx = Set key number. l = loop, s = shelf, c = card, u = unit for Large Systems. c = card, u = unit for Small Systems and Succession 1000 systems. The TN can be the same set or any other digital set in the same node. Configure the TN of the same set against the BFS key only if the Class Of Service is BFEA.

Feature operation

To control the BSFE feature from the boss set:

Activate:

1 Press the BFS boss key once. The display shows:
`PRESS BFS KEY OF SEC.`

2 Press the specific BFS secretary key to designate the secretary to filter the calls. The designated secretary's BFS key lamp winks on all sets with the default lamp status.

Note: The display on the boss set will go blank. To refresh the display, press the release key.

Deactivate:

1 Press the BFS boss key once. The display shows:
`CANCEL FILTERING?`

2 Press the BFS boss key for the second time. The feature is deactivated. The designated boss BFS key lamp turns DARK on all sets with the default lamp status.

To control the BSFE feature from the secretary set:

Activate:

1 Press the BFS boss key once The display shows
`ACTIVATE FILTERING?`

2 Press BFS boss key for the second time. This set becomes the secretary set. The display is cleared. The designated boss BFS key lamp WINKS on all sets with the default lamp status.

Deactivate:

1. Press BFS boss key for once. The display shows:
   `CANCEL FILTERING?`
2. Press the BFS boss key second time. The feature is deactivated. The display is cleared. The designated boss BFS key lamp turns DARK on all sets with the default lamp status.

To modify the BSFE from another secretary set:

1. Press the boss BFS key from another secretary set once. The display shows:
   `MODIFY FILTERING?`
2. Press the boss BFS key from the same set the second time. This secretary set becomes the new secretary filtering the calls of the boss set. The display is cleared.

Accept incoming call by boss:

1. Go offhook; press the SCR key.
2. Press the BFS boss key.

To transfer an incoming call from the secretary to the boss set:

1. Go off hook/press the SCR key to answer the ringing call.
2. Press the BFS boss key for the first time. The boss set rings.
3. The boss set answers the call.
4. Press the BFS boss key for the second time; this moves the call from the secretary set to the boss set.

The display - boss and secretary:

1. Press the Display key.
2. Press the BFS key. The set display shows the DN number of set filtering the boss calls.
3. The name and number of calls being filtered is displayed on the boss set.

   Note: When the BSFE feature is activated on the boss set, the BFS key flashes on all secretary sets associated with the boss set. This indicates the boss calls are being filtered. Each secretary can press the BFS key to display on their set. The secretary set filters the calls.

Bridging

Contents

This section contains information on the following topics:

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Feature packaging 568
Feature implementation 568
Feature operation 568

Feature description

With Bridging, the same DN can appear on up to eight single-line telephones. A maximum of five of these telephones can be equipped with ringers.

Incoming calls ring all telephones with a ringer connected and can be answered at any of the single-line telephones.

Operating parameters

A maximum of five C4A ringers are allowed on one parallel loop.

Feature interactions

Privacy

Privacy is lost when telephones are bridged. Any appearance of the DN can enter the call by going off-hook.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Busy Lamp Field

Contents

This section contains information on the following topics:

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Feature implementation 574
Feature operation 577

Feature description

When a DN is blocked due to the Attendant Blocking of Directory Number feature, the Busy Lamp Field/Enhanced Busy Lamp Field lamp corresponding to this DN displays the busy status of the DN as for ringing calls.

There are two types of Busy Lamp Field (BLF) modules.

QMT3 Lamp Field Array

The QMT3 Lamp Field Array is an add-on module for SL-1 telephones and QCW Attendant Consoles. It displays the status of a specified 150 consecutive Directory Numbers (DNs), defined in LD 15 (Standard Busy Lamp Field [SBLF]). A maximum of two Lamp Field Arrays can be supported per customer. Both Lamp Field Arrays in the customer group display status for the same 150 DNs.

Busy Lamp Field/Console Graphics Module

The Busy Lamp Field/Console Graphics Module (BLF/CGM) is an add-on module for the M1250 or M2250 Attendant Consoles. It can be configured to display the status of a specified 150 consecutive DNs (Standard Busy Lamp Field (SBLF), or all DNs, 100 at a time (Enhanced Busy Lamp Field [EBLF]). By monitoring the status, an attendant can tell a caller if the DN is busy prior to extending the call.

Enhanced Busy Lamp Field (EBLF) Array, displays the status of all DNs for a customer. The BLF/CGM displays the status of 100 DNs at a time on up to 63 M1250/M2250 Attendant Consoles. Each of the Console Graphics Modules can display a different hundreds group, while up to 20 CGMs can display the same hundreds group simultaneously.

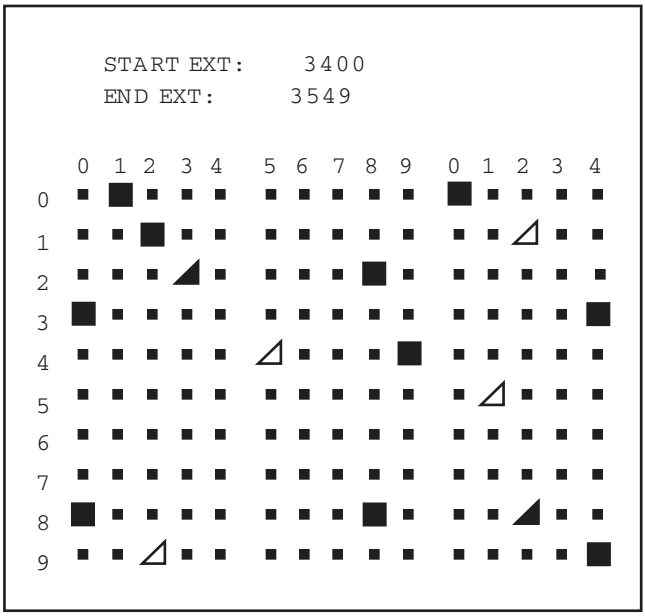
When the attendant extends a call, a hundreds group is displayed after enough digits have been entered to determine the group. After a group has been established, the BLF/CGM shows the status for each DN in that group. Figure 8 shows an example of the EBLF on the BLF/CGM.

The EBLF continues to display the status of the hundreds group until another group is determined or until the module is cleared. The display is updated whenever the status of a DN in that group changes. The BLF is cleared when the attendant dials a new series of digits or releases the call.

Figure 8 shows the Standard Busy Lamp Field (SBLF) display on the CGM. The first and last DNs in the displayed group are listed as START EXT and END EXT. The START and END EXT DNs show the hundreds group displayed. The top row on the CGM designates the tens group. The left side shows the ones group. Figure 8 shows the busy DNs to be 3403, 3408, 3410, 3421, 3482, 3488, 3494, 3500, 3543, and 3549.

Figure 9 shows a system monitored by the EBLF. Each telephone represents a busy DN, listed beneath the telephone icon. The display screen at the top of the module defines the hundreds group as 35. The CGM displays the busy DNs within that group. The larger squares represent busy telephones within the group, and the smaller squares represent idle DNs. The attendant can quickly see which telephones are busy and which are idle.

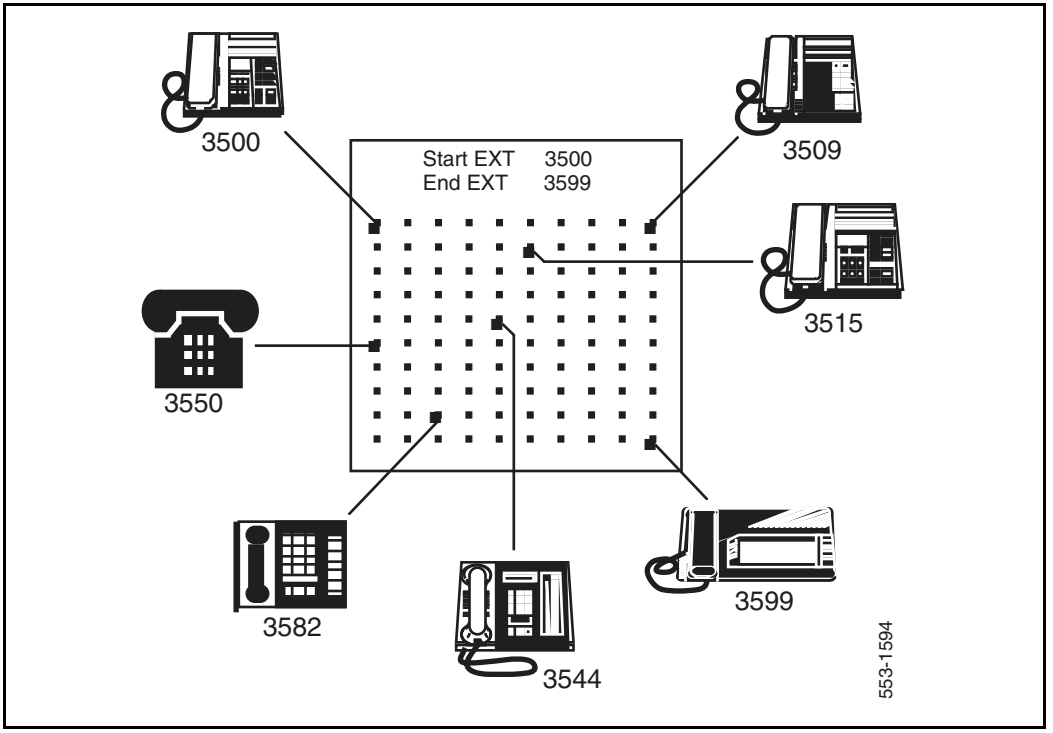
Figure 8
Standard Busy Lamp Field on the BLF/CGM



■ = idle extension
■ = busy extension
◿ = idle extension with supplementary information
◢ = busy extension with supplementary information

553-5109

Figure 9
Enhanced Busy Lamp Field monitoring (example)



Operating parameters

Enough hundreds groups must be defined to support the maximum number of telephones to be monitored. The maximum number of hundreds is 99.

The EBLF requires an M1250/M2250 Attendant Console equipped with a BLF/CGM. It does not work with the earlier Attendant Consoles using a QMT3 Lamp Field Array.

The SBLF and the EBLF are incompatible.

The EBLF supports mixed dialing plans (4, 5, 6, or 7 digits), but each hundreds group defined must be unique. For example, DNs 25XX and 25XXX cannot be configured in the same system. Any other DN group must begin with something other than 25 because, in this case, the CGM would be updated for DNs 2500 through 2599.

Only 20 Attendant Consoles can be updated for the same hundreds group simultaneously. If more than 20 consoles are monitoring the status of a single hundreds group, only the first 20 are updated. The remaining consoles display the earlier status, and an error message is output at this occurrence. (An unlimited number of consoles can be updated when they display different hundreds groups.)

When the Make Set Busy key is activated or deactivated, BLF updates only the first DN it finds on the Attendant Console. Lamp audit updates the status of subsequent DNs on the BLF.

Feature interactions

Attendant Blocking of Directory Number

When a DN is blocked due to the Attendant Blocking of DN feature, the Busy Lamp Field/Enhanced Busy Lamp Field lamp corresponding to this DN displays the busy status of the DN as for ringing calls.

Call Park

A busy lamp field can be equipped to display the status of System Park DNs.

Idle Extension Notification

When an extension that is being supervised for an Idle Extension Notification to the attendant becomes idle, it is kept busy from receiving any incoming calls. The lamp on the Attendant Console for that DN will display a busy status, according to the parameters of the Busy Lamp Field/Enhanced Busy Lamp Field feature.

It is not possible to request Idle Extension Notification if the Busy Verify feature has been activated after the Busy Verify key is pressed.

Make Set Busy

When a Make Set Busy key is activated, the Busy Lamp Field array will indicate that the first DN only on that set is busy.

Feature packaging

- Busy Lamp Field Array (BLFA) is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Define the Busy Lamp Field/Console Graphics Module options in the Customer Data Block.

2 LD 12 – Identify which attendant consoles have Enhanced Busy Lamp Field on the BLF/CGM.

3 LD 10 – Activate DN hundreds groups for EBLF for each DN within each hundreds group.

4 LD 11 – Activate DN hundreds groups for EBLF for each DN within each hundreds group.

LD 15 – Define the Busy Lamp Field/Console Graphics Module options in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(XLF) ILF (XBL) IBL	(Exclude) include Standard Busy Lamp Field. (Exclude) include Enhanced Busy Lamp Field.
- LFTN	l s c u c u	Lamp Field TN for first display console. Prompted only if OPT = ILF. For Small Systems and Succession 1000 systems.
- LFTN	 l s c u c u	Lamp Field TN for second display console. Secondary TN if this is the Attendant Console. For Large Systems For Small Systems and Succession 1000 systems
- LFFD	xxx…x	First DN for the Lamp Field Array for ILF; last two digits of the first DN must be 00. First DN must start on even 100 (for example, 3400 is acceptable, but 3450 is not).

LD 12 – Identify which attendant consoles have Enhanced Busy Lamp Field on the BLF/CGM.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
EBLF	(BLFD) BLFA	(Deny) allow Enhanced Busy Lamp Field.

Note: When the BLF is configured before the telephones are programmed, the procedures in LD 10 and LD 11 are not required. As an alternative to reentering data when the BLF is configured after the telephones, a SYSLOAD associates the DN with the Hundreds Group (HGRP).

LD 10 – Activate DN hundreds groups for EBLF for each DN within each hundreds group.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DN	xxx…x	Reenter Directory Number (no change necessary).

LD 11 – Activate DN hundreds groups for EBLF for each DN within each hundreds group.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx aaa yyy…y	Reassign Directory Number (no change necessary), where: xx = key number aaa = DN type, and yyy...y = Directory Number.

Feature operation

To display the status of extensions on the BLF/CGM (attendant), follow these steps:

1. Press the **SHIFT** key, then the conf. 6/BLF key.
   The console is in the BLF mode.

2. Press the **Mode** key .
   The BLF/CGM screen displays the main menu.

3. Dial 0 (zero).
   The BLF/CGM displays the SBLF or the EBLF, depending on which option is configured in the system software.

Busy Tone Detection for Asia Pacific and CALA

Contents

This section contains information on the following topics:

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Feature operation 586

Feature description

The Busy Tone Detection feature for Asia Pacific and CALA uses the Digital Signaling Processor Universal Trunk (DXUT) card. This card is based on the Extended Universal Trunk card (EXUT) and allows for the following two capabilities:

- Flexible Busy Tone Detection
- Automatic Balance Impedance (AUTO_BIMP in Overlay 14)

The Flexible Busy Tone Detection functionality of this trunk card allows the system to recognize busy tones sent from a Public Exchange/Central Office. Busy Tone Detection permits disconnect supervision for Loop Start Central Office (CO) trunks. The Central Office provides busy tone to the last party involved in a call. The system detects this busy tone and disconnects the call.

Busy Tone Detection features are utilized in countries where tone detection is the only method for the system to detect far end disconnection.

The Busy Tone Detection feature for Asia Pacific and CALA uses the NT5D31 Digital Signaling Processor (DSP) Universal Trunk (DXUT) card. This card is based on the Extended Universal Trunk card (EXUT) and is configured in software as an EXUT card. However, the DXUT card has flexible busy tone detection provided by a Digital Signal Processor (DSP). The DXUT card also has tone detection intelligence that allows it to accurately differentiate between different disconnect tones sent by a Public Exchange/Central Office.

The DXUT card has programmable Busy Tone Detection characteristics which include:

- Cadence
- Incoming or Incoming and Outgoing call direction
- Tone Frequencies
- Tone Bandwidth
- Tone Levels

Tones are detected according to the parameters configured in Overlay 97.

When a trunk card does not support the Busy Tone Detection feature, it can still be configured in software; although, the hardware does not recognize the new Busy Tone messages. The DXUT messages are ignored by the old hardware.The existing hardware is still operational since the Busy Tone feature still supports the older hardware. Old messages are sent for backwards compatibility but are not resent to define frequency criteria.

The Automatic Balance Impedance (AUTO_BIMP) functionalities of the DXUT card enhance the Transhybrid Loss matching capability. The automatic balancing is performed by the Digital Signal Processor (DSP) when checking the reflections from the transmission line. When the software sends an AUTO_BIMP message to the DXUT card, the DSP generates a test tone and measures the amount of signal being reflected. The DSP then internally adjusts the balance network, in the codec, for the best Transhybrid loss.

Operating parameters

The Busy Tone Detection feature for Asia Pacific and CALA requires the DXUT card. The DXUT card requires busy tone detection data to be downloaded prior to activating this feature.

The AUTO_BIMP functionalities of this feature are not supported in the Digital Signaling Processor Universal Trunk (DXUT) card NT5D31 hardware.

Direct Inward Dialing (DID) trunks do not require busy tone supervision, since the Public Exchange/Central Office seizes the system trunk by closing the transmission loop. Far end trunk release is accomplished when the Public Exchange/Central Office opens the circuit.

Japan trunk cards, the Extended Universal Trunk card for Japan (XUTJ) and the Enhanced Extended Universal Trunk card for Japan (EXUTJ), do not support this feature. The DXUT card is not supported in Japan.

The system disconnects a call when a busy tone is detected on an incoming trunk. If the caller on the far end causes a busy tone to be generated, the call is disconnected, regardless of whether or not disconnection was intended. As an example, when a caller connected to a Public Exchange/Central Office attempts to conference in a busy party, the system picks up this busy tone and the call is disconnected.

If any other types of tones (other than busy tone) are detected with the same cadence, frequency and level, the call is disconnected.

The Busy Tone Detection feature for Asia Pacific and CALA may not operate on conference bridges. In the scenario of Busy Tone Detection operating with a conference bridge, all of the trunks are incoming and an incoming Public Exchange/Central Office trunk disconnects from a conference. In this scenario, the disconnected trunk sends a busy tone signal to the conference bridge, and all trunks may be disconnected simultaneously.

In the event that an incoming call is connected to an external conference and two different Public Exchanges/Central Offices are sending busy tone signals at the same time, a stalemate condition may exist. When this occurs, the cadence of both busy tones may not be the same, and the resulted combination cadences may not be detected.

The DXUT card is based on the EXUT card design and is intended to operate in an EXUT- compatible Loss Planning environment. These EXUT compatible Loss Planning environments include the North American Loss Planning environment and Dynamic Loss Switching environments in certain countries.

Busy Tone characteristics are downloaded on a card basis. The Busy Tone Detection table assigned to the card is downloaded to the card when: the first trunk is configured, the card is disabled and enabled, the card is unplugged and reset, and during initialization after sysload, and when the Intelligent Peripheral Equipment package 203 is enabled.

Feature interactions

European XFCOT Support

When the XFCOT Busy Tone ID (BTID) is configured in Overlay 14 only the BTID is downloaded to the XFCOT card. The BTID is downloaded to the EXUT card when the Busy Tone Detection (BTD) package 294 is equipped.

Trunk to Trunk Connection

When the Trunk to Trunk Connection feature interacts with Busy Tone Detection for Asia Pacific and CALA, which ever feature occurs first takes precedence.

Timed Forced Disconnect

When Timed Forced Disconnect interacts with Busy Tone Detection for Asia Pacific and CALA, whichever feature occurs first takes precedence.

Feature packaging

Busy Tone Detection for Asia Pacific and CALA requires Busy Tone Detection (BTD) package 294.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 97 – Configure Busy Tone Detection (BTD) table parameters.

2 LD 16 – Configure trunk units and trunk timers in the Route Data Block.

3 LD 14 – Configure Busy Tone Supervision for a new Central Office Trunk.

Note: Once the BTD table is configured, the new trunks can be entered and the required BTD table is assigned on a card basis. The BTD table number can only be entered in for the first unit programmed on the card.

LD 97 – Configure Busy Tone Detection (BTD) table parameters.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	BTD	Busy Tone Detection.
BTDT	(0) - 7	Busy Tone Detection Table.
BCAD	(350) (350) 500 500	Busy Tone Cadence (in milliseconds). (ON cycle) (OFF cycle) (default) For Japan. The values for each cycle are 0 to 1.5 seconds (1500 ms) and are entered in milliseconds. Input values are rounded to the nearest multiple of 25 ms. If zero (0) is entered for both phases, then a continuous tone occurs.
BTDD	(BOTH) INC	Busy Tone Detection Direction: Both Incoming and outgoing calls (default). Incoming calls only.
FREQ_0	350 - 655	Frequency of Busy Tone for Frequency 0 of a dual Busy Tone Detection to be detected in Hz. Valid entries are in multiples of 5Hz.

FREQ_1	350 - 655	Frequency of Busy Tone for Frequency 1 of a dual Busy Tone Detection to be detected in Hz. Valid entries are in multiples of 5Hz. For a single busy tone FREQ_1 must be set the same as FREQ_0.
FDLT	10 - 315	Frequency Delta. FDLT gives the tolerance of the tone to be detected in +/- hertz. Valid entries are in multiples of 5Hz. For dual Busy Tone Detection on the NT5D31 card, the same maximum and minimum levels apply to both tones.
FLVL_MAX	0 - 15	Maximum Frequency Tone level to be detected. Valid entries are in multiples of 5dBm. For dual Busy Tone Detection on the NT5D31 card, the same level applies to both tones.
FLVL_MIN	20 - 35	Minimum Frequency Tone level to be detected. Valid entries are in multiples of 5dBm. For dual Busy Tone Detection on the NT5D31 card, the same level applies to both tones.

LD 16 – Configure trunk units and trunk timers in the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add a new data block to the system.
TYPE	RDB	Define a new Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	COT	Define trunk type as Central Office.
ICOG	IAO	Incoming and Outgoing trunk.
CNTL	YES	Changes to controls or timers.

NEDC	ETH	Either end control.
FEDC	ETH	Either end control.

LD 14 – Configure Busy Tone Supervision for a new Central Office Trunk.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	COT	Central Office trunk.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	EXUT	Type is IPE EXUT. This includes the DXUT. (This prompt is required only for the first unit defined on each card.)
CUST	xx	Customer number, as defined in LD 15
RTMB	0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
SIGL	LOP	Loop start level 3 signaling.
TIMP	(600) 900	Termination Impedance.
BIMP	(3COM) 3CM2 600 900	Balance Impedance. In the case of AUTO_BIMP, this BIMP value is used as a default value if an optimum AUTO_BIMP is not found or if the AUTO_BIMP test is not complete.
AUTO_BIMP	YES	Automatic Balance Impedance is set according to transmission line parameters. NO = default for new trunks.
SUPN	YES	Answer and disconnect supervision required.

-STYP		Supervision Type.
	PIP	Polarity Insensitive Pack.
	BTS	Busy Tone Supervision.
	PIP BTS	Both options.
BTDT	(0)-7	Busy Tone Detection Table number configured in LD 97. (This prompt is required only for the first unit defined on each card.)
CLS	(DIP)	Dial Pulse.
	DTN	Digitone.

Feature operation

No specific operating procedures are required to use this feature.

Busy Tone Detection for Japan

Contents

This section contains information on the following topics:

Feature description 587
Operating parameters 588
Feature interactions 589
Feature packaging 589
Feature implementation 589
Feature operation 592

Feature description

In many countries, Central Office loop start trunks are not supervised. This can lead to difficulties for incoming calls to the system that require disconnect supervision to operate properly. Through a modification to the tone detector, this feature allows the system to perform disconnect supervision through the recognition of a busy tone sent by the Public Exchange/Central Office.

Busy Tone Detection for Japan allows a technician to enter the characteristics of the busy tone tables in LD 97. When these characteristics are programmed, the information is downloaded to the system during call processing. When a busy tone is detected, the trunk sends a message to the system software to disconnect the call and free the trunk for other uses.

This feature provides Japan Central Office (JCO) and Japan Direct Inward Dialing (JDID) trunks with Busy Tone Detection (BTD) capability through trunk supervision.

Operating parameters

The feature is applicable to all systems.

Busy Tone Detection for Japan requires the Enhanced Extended Universal Trunk Card for Japan (EXUTJ).

This feature requires a busy tone from the Public Exchange/Central Office.

The system disconnects any call if a busy tone is detected on the incoming trunk. If called party causes a busy tone to be generated, the call disconnects whether intended or not. As an example, this may happen if a Central Office user tries to conference in a busy party. The busy tone is detected by the circuit switched network trunk and the call disconnects.

If another tone is configured similar to the Busy Tone (frequency + or - 30 Hz and cadence within + or - 100 ms), the busy tone detector is interpreted as a busy tone and the call is disconnected. Therefore, tones should be configured so they can be interpreted correctly.

The busy tone detection characteristics are downloaded on a card basis only. All units on the trunk card must go to the Central Office that produces the same Busy Tone cadence.

To modify the busy tone detection table assigned to a trunk card, all trunks on that card must be removed initially from the software (LD 14). It is recommended that all Central Office loop start trunk units be on the same card and configured in the same route.

500/2500 Line Disconnect Supervision is supported by this feature.

If the trunk card is not designed to support the Busy Tone Detection (BTD) feature, BTD can still be configured in the software. However, no feedback is given to the technician that a discrepancy exists between the software and hardware configuration.

Feature interactions

Timed Forced Disconnect

Busy Tone Detection for Japan activates a timer to start once a Central Office (as well as other types of trunks) has been seized. After this timer expires, the trunk is forced to disconnect. BTD does not impact this timer; however, whichever timer occurs first will prevail.

Trunk to Trunk Connection

Busy Tone Detection for Japan does not impact the Trunk to Trunk Connection feature. However, which ever occurs first prevails.

Feature packaging

Busy Tone Detection for Japan is Busy Tone Detection (BTD) package 294.

The following packages are also required:

- Japan Central Office Trunk (JPN) package 97
- Intelligent Peripheral Equipment (XPE) package 203

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 97 – Assign Tone Characteristics to Busy Tone Detection Tables.

2 LD 14 – Assign Busy Tone Detection to Central Office (CO), Foreign Exchange (FEX) and WATS Trunks.

3 LD 14 – Assign Busy Tone Detection to Direct Inward Dialing (DID) Trunks.

LD 97 – Assign Tone Characteristics to Busy Tone Detection Tables.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	BTD	Busy Tone Detection data block.
BTDT	(0)-7 X1-X7	Busy Tone Detection table. Table 0 can be changed but cannot be removed. Table 0 should always exist (when the BTD package is equipped) and is initialized to the default value for Japan. When creating alternate tables, table 0's values are used to fill the table and these can be changed. Enter X in front of the table number to remove the table.
BCAD	500 500 (ph1 ph2)	Busy Tone Cadence (on and off phase length during the cycle can be entered). ph1 is the ON cycle and ph2 is the OFF cycle. The values for each phase can be 0 to 1.5 seconds (1500 ms) and are entered as ms. The input values are rounded to the nearest multiple of 25 ms. Entering all 0s indicates continuous tone. A tone is deemed continuous if it lasts for at least 3.2 seconds. The smallest cadence is 50 ms even though 25 ms can be entered.
BTDD	(BOTH) INC	Busy Tone Detection Direction. BOTH = both incoming and outgoing calls INC = incoming calls only

LD 14 – Assign Busy Tone Detection to Central Office (CO), Foreign Exchange (FEX) and WATS Trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	COT	Central Office Trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XUT	Enhanced Extended Universal Trunk.
...		
SIGL	LOP	Loop start signaling.
...		
SUPN	YES	Answer and disconnect supervision required. If SUPN = YES, then the values stored in supervision type (STYP prompt) are initialized and only the current entered values are saved. Therefore, complete supervision is required every time through this branch.
- STYP	xxx	Trunk supervision type where xxx is: PIP = Polarity Insensitive JCO = Japan Central Office BTS = Busy Tone Supervision
...		
BTDT	(0)-7	Busy Tone Detection Table. This table must be defined in LD 97.

LD 14 – Assign Busy Tone Detection to Direct Inward Dialing (DID) Trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DID	Direct Inward Dialing Trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XUT	Enhanced Extended Universal Trunk.
...		
SIGL	LOP	Loop start signaling.
...		
SUPN	YES	Supervision. This response is automatically prompted YES for DID LOP.
- STYP	xxxx	Trunk supervision type where xxxx is: JDID = Japan DID. When XTRK = XUT and <CR> is entered STYP default to JDID. JDID BTS = Busy Tone Supervision and JDID (XUT only). When XTRK = XUT and BTS is entered STYP defaults to JDID BTS.
...		
BTDT	(0)-7	Busy Tone Detection Table. This table must be defined in LD 97.

Feature operation

No specific operating procedures are required to use this feature.

Busy Verify on Calling Party Control Calls

Contents

This section contains information on the following topics:

Feature description 593
Operating parameters 594
Feature interactions 594
Feature packaging 594
Feature implementation 595
Feature operation 595

Feature description

This enhancement to the Busy Verify feature changes the way in which a local attendant and toll attendant, and Network Attendant Service attendant are able to Busy Verify, Barge-In, and Break-In to a station that is connected to a trunk on a route that has Calling Party Control (CGPC) set to YES.

Table 29
Busy verify on calling party control calls operation for a local call

	Busy Verify	Barge-In	Break-In
Local attendant	Yes	Yes	Yes
Toll attendant	–	–	Yes
NAS attendant	–	–	Yes

Table 30
Title: Busy verify on calling party control calls operation for a toll call

	Busy Verify	Barge-In	Break-In
Local attendant	No	No	No
Toll attendant	–	–	No
NAS attendant	–	–	No

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Break-In

Local Attendant Break-In will be temporarily denied if the desired party is already in a toll operator Break-In conference or on a Special Service call, or awaiting the Special Operator signal. Local attendant/toll operator Break-In will be temporarily denied if the desired party is established on an incoming toll call.

Network Attendant Services (NAS)

A NAS attendant is not allowed to Busy Verify to a station on a different node, or Barge-In to a trunk on a different node. A NAS attendant is allowed to Break-In to a station on a different node, if the incoming trunk on the route is not a toll call. NAS attendant Break-In will be temporarily denied if the desired party is already on a toll call, a toll operator Break-In conference, or a Special Service call, or awaiting the Special Operator signal.

Feature packaging

Busy Verify on Calling Party Control Calls requires Operator Call Back (OPCB) package 126.

Feature implementation

No change to existing configuration is required for the Busy Verify on Calling Party Control Calls feature.

Feature operation

See the following feature descriptions contained within this document.

- “Attendant Busy Verify” on page 275
- “Attendant Barge-In” on page 231
- “Attendant Break-In” on page 237

Call Detail Recording

Call Detail Recording (CDR) records information about selected calls for accounting purposes. For each call, CDR identifies the calling and called parties and notes the time and duration of the call. A record describing the complete call is output by the system when the call is terminated. The following five recording options are available and can be specified by the customer in any combination for each trunk route:

- all outgoing calls
- all outgoing toll calls
- outgoing answered calls
- outgoing answered toll calls
- all incoming calls

For outgoing calls, all calls seizing a trunk in that route are recorded from the time of trunk seizure, no matter how long or short the call is. If answer supervision is allowed on the system, calls placed over tandem TIE trunks are billed from the time the call is answered. The caller is not charged the time it takes for dialing, digit outpulsing, or ringing.

For incoming calls, all calls answered by a telephone or Attendant Console are recorded.

Three output options are available:

- System terminal: CDR system terminal (CTY)
  Information is output in ASCII serial format suitable for a system terminal or equivalent device.

- Magnetic tape: CDR Data Link (CLNK)
  Information is output in binary format to a QCA11 CDR machine for downstream processing.
- Both system terminal and magnetic tape.

The system provides access to as many as 16 input/output ports, which can include any combination of designated CDR system terminal (CTY) or CDR Data Link (CLNK) ports. Because each customer on a system can access multiple CDR ports, system terminal and magnetic tape CDR recording machines can be used at the same time for the same customer.

For further information on CDR, please refer to *Call Detail Recording: Description and Formats* (553-3001-350).

Call Forward All Calls

Contents

This section contains information on the following topics:

Feature description 599
Operating parameters 600
Feature interactions 601
Feature packaging 612
Feature implementation 613
Feature operation 616

Feature description

Call Forward All Calls (CFW) automatically forwards incoming calls to another destination, within or outside the system. Only calls to the Prime DN or any single-appearance DN on the telephone are forwarded. Outgoing calls can still be placed from the telephone when Call Forward is active.

Call Forward All Calls can be selectively activated depending on the source of the originating party. With the Internal Call Forward (ICF) feature, the user can cause only internal calls to be forwarded. The Call Forward Reminder Tone (CFRT) presents special dial tones on analog (500/2500 type) telephones with CFW active. One tone indicates that CFW is active; a second indicates that there is a message waiting for the telephone with CFW active.

Call Forward All Calls, as well as Internal Call Forward, is assigned on a per-telephone basis. Meridian 1 proprietary telephones must be equipped with separate key/lamp pairs to allow the activation and deactivation of each feature. Customers can specify the length of the destination number in LD 11. Options are 4, 8, 12, 16, 20, or 23 digits. If you enter another number for the length, the system rounds to the nearest acceptable choice. The default is 16 digits.

When you use Multiple Appearance DNs (MADNs), call redirection is determined based on the Terminal Number (TN) order in your DN block. To determine the TN order, print the DN block from LD 20 or LD 22 (TYPE = DNB). When a call comes in to an MADN, the system begins a search to determine how the call will be handled. Using the TN list you printed, the system performs the following search, beginning at the bottom of the TN list, and working up.

1 Searches for the first Prime DN appearance of the MADN with Call Forward All Calls activated.

2 If there are no Prime DN appearances, the Call Forward All Calls cannot be activated.

Note: The search does not necessarily determine the lowest numerical TN. The search starts at the bottom of the TN list.

Operating parameters

The forwarding of a call depends on the access restrictions assigned to the telephones and the trunks involved in the call. If call forwarding results in a connection that is not permitted by the assigned access restrictions, the incoming call is not forwarded.

The customer can specify which telephone determines the successful completion of the call: the originating telephone or the forwarding telephone.

Internal Call Forward requires a programmable feature key. Therefore, Internal Call Forward is not supported on BRI telephones.

LD 17 CFWS allows telephones to have their CFW status saved as part of the data dump routine and then reinstated following a SYSLOAD. For more information, refer to the Call Forward Save feature RL.

Call Forward Reminder Tone does not apply to telephones such as the SL-1 that have a visual indication of active CFW status.

The Reciprocal Call Forward All Calls option prevents the situation whereby an infinite loop is caused in a network-wide Call Forward configuration resulting from telephone A being call forwarded (all calls) to telephone B at another node, which in turn has been call forwarded back to telephone A. A check is provided using the Flexible Orbiting Prevention Timer (FOPT), which prohibits any telephone from call forwarding more than one call off node for a period of 14 seconds.

The Flexible Orbiting Prevention Timer, previously fixed at 14 seconds, can be set during Service Change from 0 to 30 seconds (even numbers only). If a value of 0 is defined, then Orbit Prevention is disabled and call forwarding is not inhibited in any way.

The Orbit Prevention protection, however, does not extend to all potential orbiting situations. Improperly engineered networking or multiple switching arrangements can produce orbiting.

Feature interactions

Advice of Charge for EuroISDN

Calls charged with Advice of Charge that are either transferred, extended or redirected to another set using Call Forward All Calls are charged against the last set that answers the call and the controlling set releases. Additionally, the last party that transfers or forwards a call to an ISDN Central Office trunk is charged for both connections.

Attendant Alternative Answering

Call Forward All Calls takes precedence over all other Call Forwarding features for a particular telephone. Calls forwarded by Attendant Alternative Answering (AAA) are subject to the Call Forwarding conditions on the AAA DN.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override Call Forward All Calls. If the dialed DN of the set is idle, the DN can be blocked; if the DN is busy, busy tone will be heard.

Attendant Break-In

The attendant can override call forwarding on a destination DN by pressing the Break-In key before dialing the destination DN. The attendant may not apply Camp-On to a telephone with Call Forward active.

Attendant Break-In to Inquiry Calls

The operation of Call Forward All Calls is overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Call Forward All Calls that may be applied to the set.

Attendant Busy Verify

If the DN is call forwarded to the Attendant Console, the attendant will receive a click followed by silence.

Attendant Overflow Position

If the telephone assigned an Attendant Overflow DN has activated the Call Forward All Calls feature, overflow calls are not rerouted to the telephone. If a analog (500/2500 type) telephone is forwarded, AOP is canceled.

Call Forward Destination Deactivation

If a user's call forwarded Directory Number (DN) is defined as DN of Set B and set A dials the CFW FFC to activate call forward, then Set A gets forwarded to Set B. Set B can deactivate CFW on set A by dialing the Call Forward Destination Deactivation (CFDD) FFC.

Call Detail Recording on Redirected Incoming Calls

The Call Detail Recording on Redirected Incoming Calls feature does not affect how the Call Forward All Calls feature operates; however, it does provide information about the answering party in the Call Detail Recording ID field if incoming calls have been redirected by any one of these features.

Call Forward and Busy Status

Call Forward All Calls must be assigned to Party A's telephone to enable the Call Forward Status function, which allows party B to monitor and alter the Call Forward state of party A's telephone.

Call Forward by Call Type

If a call is unanswered at the forwarded DN, the telephone that has Call Forward All Calls activated is checked for the Class of Service and the call forward DN. If a chain of call forwarding occurs, the Class of Service and the forward DN for Call Forward No Answer are obtained from the first telephone in the chain. This applies when FDN and HNT have been specified for Call Forward No Answer at the customer level.

Call Forward Destination Deactivation

If a user's call forwarded Directory Number (DN) is defined as DN of Set B and set A dials the CFW FFC to activate call forward, then Set A gets forwarded to Set B. Set B can deactivate CFW on set A by dialing the CFDD deactivation FFC.

Call Forward External Deny

This feature overrides other Call Forward All Calls parameters. For example, if Call Forward to Trunk Access Code (CFTA) is allowed for the customer, but Call Forward External Deny (CFXD) is enabled for the telephone, CFXD takes precedence and call forwarding to a trunk access code is denied.

Call Forward/Hunt Override Via Flexible Feature Code

The Call Forward All Calls feature is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Call Forward, Internal Calls

If Call Forward Reminder Tone Allowed (CFRA) is in effect, whenever an analog (500/2500 type) telephone with Internal Call Forward active goes off hook to originate a call, the telephone sounds the reminder tone. The reminder tone is either Call Forward Dial Tone (CFDT) or Call Forward/Message Waiting Dial Tone (CFMW).

Call Forward No Answer

Suppose that party A calls party B, and party B has programmed Call Forward All Calls to party C. Flexible Call Forward No Answer will forward a No Answer call at party C to the FDN associated with party B, the dialed DN.

Call Forward No Answer, Second Level

Both first and Second Level Call Forward No Answer use the final (ringing) telephone in the chain to obtain Class of Service and forwarding DN information.

Call Forward Save on SYSLOAD

The Call Forward status of each telephone can be saved as part of the data dump routine and reinstated following a SYSLOAD operation.

Call Page Network Wide

Call Page Network Wide (PAGENET) does not block a station set from being programmed to Call Forward All Calls to an external Paging trunk. At call termination time, calls that are forwarded to an external PAGENET uncontrolled trunk are not blocked. However, calls forwarded to an external PAGENET controlled trunk are given access denied intercept treatment at the Paging node.

Call Redirection by Time of Day

Call Forward All Calls has precedence over Call Redirection by Time of Day.

Calling Party Name Display Denied

During a Call Forward or Call Transfer, the calling party digits and forwarding/transferring party digits are displayed on the terminating set. This display is allowed or denied depending on the Class of Service of the calling set and the forwarding/transferring set. The name of the forwarding/ transferring set is not displayed on the calling and terminating set.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Call Forward Busy
Call Waiting
Call Waiting Redirection
Camp-On
Camp-On, Station

Call Forward All Calls has precedence over Call Forward Busy, Call Waiting, Call Waiting Redirection, Camp-On and Station Camp-On.

China – Attendant Monitor

If an attendant attempts to monitor a DN which is Call Forwarded and is idle, idle DN treatment is given.

China – Flexible Feature Codes - Customer Call Forward

Enhanced Flexible Feature Codes - Customer Call Forward

When Customer Call Forward (CCFW) is active CFWAC cannot be activated by Flexible Feature Code, but can be activated by SPRE. When CFWAC is active, CCFW cannot be activated.

CCFW can be deactivated by deactivating CFWAC. CFWAC can only be deactivated by the CCFD FFC if the current CFW DN is the same as the current CCFW DN.

China – Flexible Feature Codes - Outgoing Call Barring

Enhanced Flexible Feature - Outgoing Call Barring

When a set with Outgoing Call Barring active activates CFWAC with a new CFW DN, the CFW DN is tested against the current barring level. If the DN is not allowed to be dialed, it can also not be used as a Call Forward DN. This is to prevent a set from forwarding to a barred DN 5and then dialing its own DN to bypass the restrictions.

China – Toll Call Loss Plan

Toll pad switching is also provided after call forwarding has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Conference

On analog (500/2500 type) telephones, Call Forward All Calls can be activated or canceled during a conference call.

Display of Calling Party Denied

When a set activates any of the call forwarding features, the displays given on the calling set and the terminating set are in accordance with the Class of Service of the sets involved in the call.

If the terminating set has Dialed Name Display Denied (DNDD), the display on the terminating set reflects the name and number of the calling party and the name and the number of the forwarding set.

If the terminating set has Dialed Name Display Allowed (DNDA), the display on the terminating set reflects the number of the calling party and the name and number of the forwarding set. In both cases, the terminating set's display is in accordance with the DPD Class of Service options of the calling and forwarding sets.

For a MCDN ISDN call, the calling party's Calling Line Identification (CLID) is replaced with the ISDN route access code (ACOD) and the route member number, and the calling party's name is replaced by a string of four Xs (X X X X).

The display given on the calling set of an internal call, which has been forwarded to a set within the same switch, includes the name and number of the terminating set along with the number of the forwarding set. If the DPD Class of Service options, which are specified for the terminating set, indicate that the display of the name and number of the terminating set be denied, then on the calling set, the name of the terminating set is replaced by a string of four Xs (X X X X). The number is replaced by dashes (- - - -). If the number of the terminating set is blocked from being displayed on the calling set, the number of the forwarding set is also blocked from being displayed on the calling set, regardless of the DPD Class of Service options of the forwarding set. Conversely, if the display of the terminating set's number is allowed in the calling set, then the number of the forwarding set is also displayed on the calling set, irrespective of the DPD Class of Service options of the forwarding set.

Do Not Disturb

If activated, Call Forward All Calls will take precedence over Do Not Disturb busy indication.

DPNSS1 Diversion

Call Forward All Calls on unanswered calls are activated in the following order: Call Forward All Calls, Message Waiting, Call Forward No Answer, Slow Answer Recall. For busy sets the order is: Call Forward All Calls, Hunting, Calling Waiting/Camp On, Message Waiting Busy Forward, Call Forward Busy.

Electronic Lock Network Wide/Electronic Lock on Private Lines

For Call Forwarding, the COS and NCOS used for the forwarding call can be taken from either the forwarding set or from the forwarded set, depending on the option defined in the Customer Data Block.

For example, set B call forwards all calls to an external trunk. Set A calls set B. If OPT = CFF in LD 15 (Call Forward forwarded to party's COS and NCOS), the COS and NCOS of set B are used for forwarding the call to the trunk. If OPT = CFO (Call Forward originating party's COS and NCOS), the COS and NCOS of set A are used for forwarding the call to the trunk.

Flexible Feature Code Boss Secretarial Filtering

Although Call Forward All Calls and Flexible Feature Code Boss Secretarial Filtering can be equipped on the same set, they cannot both be active at the same time. There is no precedence of one over the other; it is not possible to activate one if the other is active on the set.

Flexible Feature Codes

When Flexible Feature Codes (FFC) are configured for a customer, #1 automatically becomes the FFC DN for both Call Forward Activate (CFWA) and Call Forward Deactivate (CFWD). When the same DN is used for both CFWA and CFWD, FFC toggles the call forward activated/deactivated state of the telephone. When call forward is activated for a telephone, entering #1 automatically deactivates call forward, no matter what follows #1. When call forward is deactivated for a telephone, the result of entering #1 depends on what follows #1:

- If the telephone goes on hook immediately, Call Forward is activated for the telephone to its previous Call Forward number.
- If a valid DN is entered after #1, Call Forward is activated for the telephone to that valid DN.
- If an invalid DN is entered after #1, Call Forward remains deactivated for the telephone.

Flexible Voice/Data Terminal Number

Voice calls directed to a dynamic voice/data Terminal Number are forwarded, if either of these features are enabled. Data calls, to a dynamic voice/data TN, are not forwarded.

Group Call

A Group Call to a telephone with Call Forward active is forwarded one step only. The Call Forward number must be a valid DN.

Group Hunt

When Group Hunting attempts to terminate on a DN which has CFW All Calls active, it will continue with the next DN in the group if the attempted DN is busy, or if the DN is idle and the response to the Call Forward Ignore (CFWI) prompt in LD 57 is "NO". If the attempted DN is idle and the response to the CFWI prompt in LD 57 is "YES", then Group Hunting will terminate and the stations associated with the DN will be rung.

Hunting

Call Forward All Calls takes precedence over Hunting.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an Intercept Computer (ICP) position, a redirected message identification "50" is sent to the ICP computer, when the call is answered.

Idle Extension Notification

When an extension that is being supervised for Idle Extension Notification becomes idle, it has the ability to make outgoing calls. If Call Forward All Calls or Intercept Call Forward are activated at the extension before the attendant presses the SACP key to ring that extension, the attendant's call will be forwarded to the Call Forward destination. The attendant display will show both the call forward DN, as well as the original extension's DN.

If the Call Forward DN is busy, SACP can be activated towards the Call Forward DN, if all the requirements for allowing Idle Extension Notification are met by this DN.

Incoming Call Indicator Enhancement

When a Direct Inward Dialing (DID) call to station that is unrestricted from receiving DID calls (UDI Class of Service) is forwarded to a UDI station due to Call Forward All Calls or Call Forward Busy, the call is RDI-intercepted to the attendant. The attendant display shows the DN of the dialed party.

If the call has been forwarded to the attendant, the Call Forward All Calls/Call Forward Busy ICI lights up, and not the RDI-intercept ICI.

ISDN QSIG Call Completion

When the Call Forward feature is activated on a local basis and an incoming Call Completion request is received, the Call Completion request is registered against the forwarded DN.

ISDN QSIG Name Display

When an incoming QSIG call, with name display presentation allowed Name Display, is forwarded locally, the calling party's name information is displayed on the forwarding set. With presentation restriction, the calling party's name information is not displayed to the destination set.

Make Set Busy

Call Forward All Calls takes precedence over Make Set Busy.

Message Registration

The party that originates a call is charged. The charge cannot be moved to another party using Call Forward All Calls.

Multi-Party Operations

A set which has activated Call Forward All Calls can still initiate calls and become the controlling party of a consultation connection. In this case, if the set mis-operates, then Multi-Party operations while re-ringing the controlling party as a part of misoperation recovery ignores the Call Forward All Calls indication present on the controlling party.

Multiple Appearance Directory Number Redirection Prime

Multiple Appearance Directory Number Redirection Prime (MARP) affects how call redirection operation is defined. Refer to the MARP module in this document for details.

Network Intercom (Hot Type D and Hot Type 1 Enhancements)

Hot Type I calls respect or override all kinds of Call Forward features (Busy, No Answer, All Calls, Internal, etc.) according to per-set definitions. If Call Forward is respected, the call becomes a normally dialed call and the originator will receive the appropriate indication on their display.

Network Individual Do Not Disturb

Call Forward All Calls takes precedence over Do Not Disturb Individual (DNDI) treatment.

Night Service Enhancements

Any call which has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Paging

Calls that originate on a TIE trunk to a telephone that is redirected to a paging route are blocked.

Periodic Pulse Metering

Metered calls transferred or extended from one station to another using the Call Forward All Calls feature are charged against the last station at which the call is answered as the controlling station releases. The last party to forward a call onto a metered Periodic Pulse Metering trunk is charged.

Phantom Terminal Numbers (TNs)

Call Forward All Calls is used in conjunction with Remote Call Forward (RCFW) to redirect incoming calls from a phantom TN/DN to a valid DN.

Call Forward cannot be overridden on phantom TNs. Overflow tone occurs if an override is attempted.

Recorded Announcement for Calls Diverted to External Trunks

If a call is forwarded to an outgoing external Central Office (CO) route with the Recorded Announcement for Calls Diverted to External Trunks (RANX) flag set, RANX is activated.

Recovery on Misoperation of Attendant Console

Call Forward takes precedence over the Misoperation feature.

Ring Again on No Answer

If an unanswered call is forwarded to another station by Call Forward All Calls, Ring Again on No Answer is applied to the originally dialed station.

Special prefix SPRE access codes

SPRE access codes cannot be used as CFW DNs. If an attempt is made to program SPRE access code as a CFW DN, the overflow tone is given at the time of CFW activation.

Total Redirection Count

Call Forward All Calls redirection is limited to the value defined in the Total Redirection Count limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

Trunk Access Codes

There is an option in LD 15 to allow or disallow telephones to program Call Forward All Calls to a Trunk Access Code. See “Call Forward External Deny” on page 657.

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

User Selectable Call Redirection

When CFW redirects a call from telephone A to telephone B, and telephone B does not answer, the Ringing Cycle Options of telephone B determines how long it rings. After the designated number of rings, the Flexible Call Forward No Answer of telephone A redirects the call.

Feature packaging

Internal Call Forward requires the 500 Set Dial Access to Features (SS5) package 73 for analog (500/2500 type) telephones, and the Flexible Feature Codes (FFC) package 139. Call Forward Reminder Tone is packaged with the Call Forward All Calls feature.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Define Class of Service for Call Forward All Calls.

2 LD 10 – Add/change Call Forward All Calls and Internal Call Forward for analog (500/2500 type) telephones.

3 LD 11 – Add/change Call Forward All Calls and Internal Call Forward for Meridian 1 proprietary telephones.

4 LD 57 – Add/change Flexible Feature Codes for Internal Call Forward.

On an analog (500/2500 type) telephone, the user accesses the Call Forward All Calls and Internal Call Forward features by dialing either the SPRE plus the feature code, or the appropriate Flexible Feature Codes (FFCs). On a digital telephone, the user accesses each feature using its feature key.

LD 15 – Define Class of Service for Call Forward All Calls.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Call Redirection
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(CFO) CFF	(CFO) = Calling party Class of Service is active during Call Forward All Calls. CFF = Forwarding party Class of Service is active during Call Forward All Calls.
	(PVCA) PVCD	Prevention of reciprocal Call Forward (allowed) or denied.

- CFTA	(NO) YES	(Deny) allow telephones to Call Forward All Calls to a Trunk Access Code.
- OPT	(CFRD) CFRA	Call Forward Reminder Tone (denied) allowed for analog (500/2500 type) telephones.

Note: In LD 56, the XCT Tone Code (XCAD) default value is set to 0. If the XCAD prompt is not changed in LD 56, an Analog 500-type set will not receive Call Forward Reminder Tone (CFRT) even if it has been enabled in LD 15 (OPT = CRFA).

LD 10 – Add/change Call Forward All Calls and Internal Call Forward for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FTR	CFW xx yyyy...y	Allow Call Forward All Calls, where: xx = maximum number of digits in the CFW DN. Valid entries are any integer in the range of (4)-31. yyyy = number where calls are forwarded. ***Note:*** YYYY cannot be entered from the maintenance terminal. When the telephone information is printed in LD 20, yyyy shows the call forward number.
FTR	ICF xx	Allow Internal Call Forward, where: xx = maximum number of digits in the Forward DN. Valid entries are any integer in the range of (4)-31.

LD 11 – Add/change Call Forward All Calls and Internal Call Forward for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx CFW yy zzzz...z	Define Call Forward All Calls, where: xx = key number; M2317 must use key 22 yy = maximum number of digits in the CFW DN. Valid entries for M2317 and M3000 sets are any integer in the range of (4)-23. For all other Meridian 1 proprietary type sets, valid entries are any integer in the range of (4)-31. zzzz = number where calls are forwarded.
KEY	xx ICF yy zzzz	Define Internal Call Forward, where: xx = key number yy = maximum number of digits in the Forward DN. Valid entries are any integer in the range of (4)-31. zzzz = number where calls are forwarded.

LD 57 – Add/change Flexible Feature Codes for Internal Call Forward.

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove an FFC table.
TYPE	FFC	Flexible Feature Code.
CODE	ICFA ICFD ICFV	Access code for Internal CFW Activate. Access code for Internal CFW Deactivate. Access code for Internal CFW Verify.
ICFA	xxxx	Internal CFW Activate Code (ICFD and ICFA may share the same code).

ICFD	xxxx	Internal CFW Deactivate Code (ICFD and ICFA may share the same code).
ICFV	xxxx	Internal CFW Verify Code.

Feature operation

To forward all calls from a Meridian 1 proprietary telephone:

1 Press **Forward**.

2 Dial the number where calls are to be forwarded.

3 Press **Forward**.

To forward internal calls only from a Meridian 1 proprietary telephone:

1 Press **Internal Call Forward**.

2 Dial the number where calls are to be forwarded.

3 Press **Internal Call Forward**.

To cancel Call Forward All Calls:

- Press **Forward**.

To cancel Internal Call Forward:

- Press **Internal Call Forward**.

To forward calls from an analog (500/2500 type) telephone:

1 Lift the handset and dial SPRE 74
or lift the handset and dial #1 (2500 telephone)
or lift the handset and dial the Call Forward Allowed FFC.

2 Dial the number where calls are to be forwarded.

3 Hang up.

Note: If you deactivate Call Forward, then reactivate without changing the number, calls will be forwarded to the previously established CFW DN.

To forward internal calls from an analog (500/2500 type) telephone:

1 Lift the handset and dial SPRE 9914
or lift the handset and dial the Internal Call Forward FFC.

2 Dial the number where calls are to be forwarded.

3 Hang up.

To cancel Call Forward All Calls:

- Lift the handset and dial SPRE 74
or lift the handset and dial #1 (2500 telephone)
or lift the handset and dial the Call Forward Deny FFC.

To cancel Internal Call Forward:

- Lift the handset and dial SPRE 9914
or lift the handset and dial the Internal Call Forward Deny FFC.

Call Forward and Busy Status

Contents

The following are topics in this section:

Feature description 619
Operating parameters 621
Feature interactions 621
Feature packaging 623
Feature implementation 623
Feature operation 625

Feature description

The Call Forward and Busy Status feature was designed for an environment where Party A forwards calls to Party B for screening.

When equipped with a Busy/Forward Status (BFS) key-lamp or key-Liquid Crystal Display (LCD) pair, Party B can perform the following:

- monitor, activate, or deactivate Call Forward for Party A
- override Call Forward of Party A, in order to place a call to Party A or
- determine whether Party A is busy on a call

The BFS key-lamp or key-LCD pair serves a dual purpose. The Busy Status function indicates to Party B, using lamp or LCD state, whether Party A is busy or not. The Call Forward Status function allows Party B to monitor and alter the Call Forward state of Party A. Therefore, the BFS lamp or LCD state of Party B may indicate that Party A is in any one of the following four possible states:

- Call Forward activated and not busy
- Call Forward activated and busy
- Call Forward deactivated and not busy or
- Call Forward deactivated and busy

The BFS key also acts as an Autodial (ADL) key. To use the BFS key as an ADL key to call Party A, Party B goes off-hook and presses the BFS key for Party A.

Busy Status

The Busy Status portion of the feature indicates if Party A is busy for any of the following reasons:

- call active on a Directory Number (DN) key (SCR, SCN, MCR, or MCN)
- has Make Set Busy (MSB) activated
- has Do Not Disturb (DND) activated
- call active on a Group Call (GRC) key
- call active on a Private-line non-ringing (PVN) or Private-line Ringing (PVR) key
- Party A ringing
- if Party A is a Meridian 1 proprietary telephones and has call on Hold
- if Party A is an analog (500/2500 type) telephone and has a call on permanent Hold

Call Forward Status

The Call Forward portion allows Party B to monitor and alter the Call Forward state of Party A. Party A may be either an analog (500/2500 type) telephone or Meridian 1 proprietary telephone and must have Call Forward All Calls equipped. The Call Forward and Busy Status feature introduces a modification to the Call Forward All Calls feature functionality. The modification is activated or deactivated on a customer-wide basis by the response to OPT in LD 15. The OPT responses are “FKD” (Forward Key Denied) and the default setting “FKA” (Forward Key Allowed).

Operating parameters

Party B must be using an SL-1, M2000-series or Meridian Modular telephone. Party A may have an SL-1, M1000-series, M2000-series, M3000, Meridian Modular or 500/2500-type telephone, with Call Forward All Calls (CFAC) equipped. The operating parameters are the same as for CFAC.

A station may be monitored by a maximum of 16 other stations using the BFS key.

The monitored and monitoring stations must all belong to the same customer.

Feature interactions

Attendant and Network-Wide Remote Call Forward

When the call forward status of a BFS station is changed from a telephone or attendant-based Remote Call Forward feature, the BFS lamp(s) associated with that station are updated accordingly.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override these Call Forward and Busy Status. If the dialed DN of the set is idle, the DN can be blocked; if the DN is busy, busy tone will be heard.

Autodial

Party A can use the Busy/Forward Status key as an Autodial key to dial Party B.

Call Forward All Calls

Call Forward All Calls must be assigned to Party A's telephone to enable the Call Forward Status function, which allows Party B to monitor and alter the Call Forward state of Party A's telephone.

Call Forward/Hunt Override Via Flexible Feature Code

Call Forward and Busy Status is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Call Forward, Remote (Attendant and Network-Wide)

When the call forward status of a BFS station is changed from a telephone or attendant-based Remote Call Forward feature, the BFS lamp(s) associated with that station are updated accordingly.

Calling Party Privacy

If an incoming ISDN trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the outgoing trunk route on the tandem node also has CCP provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (that is, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Flexible Feature Code Boss Secretarial Filtering

If the secretary set is a Meridian 1 proprietary telephone, or a compact digital set, it may be equipped with a Call Forward and Busy Status (BFS) key/lamp pair, to perform the following:

- monitor the status of the Call Forward feature on a boss set
- activate/deactivate the Call Forward feature on a boss set

- monitor whether or not a boss set is busy on a call, and
- override the Call Forward All Calls feature on a boss set, in order to place a call to the boss set.

The above functions, however, can only be performed by the secretary set while it is in an unattended state, since BFS and Flexible Feature Code Boss Secretarial Filtering cannot be active simultaneously.

Network Intercom

In a Secretarial filtering scenario, the secretary's Busy/Forward Status (BFS) lamp also will reflect that the boss' set is busy if the boss is on a Hot Type I call.

Phantom Terminal Numbers (TNs)

When a user attempts to define a BFS key for a phantom TN, the system generates the following error message: "An invalid TN has been entered for the Busy/Forward Status (BFS) key."

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Respond to the OPT prompt with either FKA, the default, (Forward Key Allowed), or FKD (Forward Key Denied) to select the Call Forward All Calls mode of operation.

2 LD 11 – For each telephone set to be given a Busy/Forward Status (BFS) key, respond to the KEY prompt with 0-69 BFS lll s cc uu where 0-69 is the key number and lll s cc uu is the TN of the monitored telephone set.

3 LD 20 – This overlay is modified to print the new BFS key type and related information.

4 LD 21 – This overlay is modified to include the FKA or FKD setting as part of the OPT setting printout.

LD 15 – Respond to the OPT prompt with either FKA, the default, (Forward Key Allowed), or FKD (Forward Key Denied) to select the Call Forward All Calls mode of operation.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	RDR	Call Redirection
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
NCOS	...	
- OPT	(FKA) FKD	Forward Key (Allowed) Denied — determines whether Call Forward keys on user sets for this customer are operational.

LD 11 – For each telephone set to be given a Busy/Forward Status (BFS) key, respond to the KEY prompt with 0-69 BFS lll s cc uu where 0-69 is the key number and lll s cc uu is the TN of the monitored telephone set.

Prompt	Response	Description
REQ:	CHG NEW	Request: Modify or create data block.
TYPE:	xxxx	Type of data block.
...		

LANG	...	
KEY	0-69 BFS l s c u 0-69 BFS c u	Key number (0-69), Busy/Forward Status (BFS), Terminal Number (TN) of set to be monitored. For Large Systems For Small Systems and Succession 1000 systems

LD 20 – This overlay is modified to print the new BFS key type and related information.

Prompt	Response	Description
REQ:	PRT	Request: Print data block.
TYPE:	xxxx	Type of data block.
TN	...	

LD 21 – This overlay is modified to include the FKA or FKD setting as part of the OPT setting printout.

Prompt	Response	Description
REQ	PRT	Request: Print data block.
TYPE	CDB	Type of data block: Customer Data Block.
CUST	xx	Customer number, as defined in LD 15

Feature operation

Call Forward Status

With FKA, the default, selected Party A's Call Forward (CFW) key-lamp or key-LCD pair operation is unaffected and depressing Party B's BFS key will result in one of the following:

- If Party A does not currently have Call Forward activated, Party A has Call Forward activated to Party B's DN and Party A's CFW lamp or LCD is activated.

- If Party A already has Call Forward activated to Party B's DN, Party A has Call Forward deactivated and Party A's CFW lamp or LCD is deactivated.
- If Party A already has Call Forward activated to a DN other than Party B's, Call Forward is left as is.

With FKD selected Party A's CFW key-lamp or key-LCD pair operation is modified so that the pair is usable as an indicator only, the key is disabled, and depressing Party B's BFS key will result in one of the following:

- If Party A has Call Forward active to the DN of a BFS key equipped set other than Party B, Call Forward is left as is.
- If Party A has Call Forward active to a remote DN and Call Forward was activated by a remote FFC, the existing Call Forward DN is overridden and all new calls are forwarded to Party B's DN.
- If Party A has Call Forward active to Party B's DN, Party A's Call Forward is deactivated and Party A's CFW lamp or LCD is deactivated.
- If Party A does not have Call Forward active, Call Forward is activated to Party B's DN and Party A's CFW lamp or LCD is activated.

  Note: When the Boss set is call forwarded to one of the secretary DNs, then calling the Boss set from the secretary by using the BFS key overrides the call forward. If the secretary calls the Boss, and Boss set has been call forwarded to a DN which is not a secretary DN, then the call is forwarded.

BFS lamp or LCD states

Party B's BFS lamp or LCD reflects the status of Party A's set in terms of both the Busy or Idle and the Call Forward states. The following table gives the possible BFS lamp or LCD states for the various states Party A can be in:

Party A's set	Party A's Call Forward	
	Deactivated	Activated
Idle	Dark	Wink = Fast Flash
Busy	Lit	Flash

Where:

Dark – indicates lamp or LCD is off.

Wink – indicates lamp or LCD is winking at 60 impulses per minute (ipm) (0.875 seconds on, 0.125 seconds off).P402402

Lit – indicates lamp or LCD is on

Flash – indicates lamp or LCD is flashing at 30 ipm (0.5 seconds on, 0.5 seconds off).

Call Forward Busy

Contents

This section contains information on the following topics:

Feature description 629
Operating parameters 629
Feature interactions 630
Feature packaging 634
Feature implementation 634
Feature operation 635

Feature description

Call Forward Busy (CFB) automatically routes incoming Direct Inward Dialing (DID) calls to the Attendant Console when a telephone is busy. This capability is allowed or denied in the Class of Service (FBA/FBD) of the telephone.

Operating parameters

On incoming DID calls, Hunting takes precedence, followed by Call Waiting, then Call Forward Busy. In busy situations, the call hunts if the telephone has Hunting specified.

Feature interactions

Attendant Alternative Answering

If Call Forward Busy is allowed for the Attendant Alternative Answering (AAA) DN (and that DN is busy), a DID call is returned to the attendant and can again be eligible for AAA timing and operation.

Attendant Busy Verify

Attendant Busy Verify does not affect Call Forward Busy.

Call Detail Recording on Redirected Incoming Calls

The Call Detail Recording on Redirected Incoming Calls feature does not affect how the Call Forward Busy feature operates; however, it does provide information about the answering party in the CDR ID field if incoming calls have been redirected by any one of these features.

Call Forward All Calls

Call Forward All Calls takes precedence over Call Forward Busy.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Call Waiting for Meridian 1 proprietary telephones

If Class of Service allows CFB and Call Waiting Allowed, and the telephone has a Call Waiting key, calls do not forward to the attendant when the telephone is busy and another call is waiting.

Call Waiting for analog (500/2500 type) telephones

If a telephone has CFB and Call Waiting Allowed Class of Service, calls are forwarded to the attendant when the telephone is busy and has another call waiting.

Camp-On, Station

For DID calls only, Call Forward Busy takes precedence over Camp-On, Station.

China – Attendant Monitor

If an attendant attempts to monitor a DN which is Call Forwarded and is idle, idle DN treatment is given.

China – Toll Call Loss Plan

Toll pad switching is also provided after call forwarding has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Departmental Listed Directory Number

Call Forward Busy operates like Call Forward to 0, and are routed to any idle Attendant Console in the customer group.

Dial Access to Group Calls

Call Forward Busy cannot be applied to a Group Call.

Call Forward Busy

Calls modified by Call Forward Busy are not given Distinctive Ringing as they terminate on the Attendant Console.

Dial Access to Group Calls
Group Call

Call Forward Busy cannot be applied to Dial Access to Group Calls or Group Call.

Flexible Feature Code Boss Secretarial Filtering

Call Forward Busy to a boss set with filtering active is routed to the secretary set.

Flexible Voice/Data Terminal Number

Voice calls directed to a call processing busy dynamic voice/data TN are redirected using Call Forward Busy provided this feature is configured for the TN. Data calls to dynamic voice/data TNs are not redirected.

Group Hunt

Group Hunting has priority over the Call Forward Busy feature.

If the DN attempted for termination has FBA (Forward Busy Allowed) Class of Service, and if it is busy, then Group Hunting continues with the next DN in the group

Hot Line

Any Hot Line telephone can be assigned Call Forward Busy but it applies only to the two-way Hot Line capability.

Hunting

When a telephone is busy, an incoming call hunts only if Hunting is allowed for that telephone. If all the steps in the hunt group are busy, and Call Waiting is not allowed, the call forwards to the Attendant Console.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an Intercept Computer (ICP) position, a redirected message identification “50” is sent to the ICP computer, when the call is answered.

Incoming Call Indicator Enhancement

When a DID call to station that is unrestricted from receiving DID calls (UDI Class of Service) is forwarded to a UDI station due to Call Forward All Calls or Call Forward Busy, the call is RDI-intercepted to the attendant. The attendant display shows the DN of the dialed party.

If the call has been forwarded to the attendant, the Call Forward All Calls/Call Forward Busy ICI lights up, and not the RDI-intercept ICI.

Lockout, DID Second Degree Busy, and MFE Signaling Treatments

Call Forward Busy takes precedence over lockout and second degree busy.

Network Intercom

Hot Type I calls respect or override all kinds of Call Forward features (Busy, No Answer, All Calls, Internal, etc.) according to per-set definitions. If Call Forward is respected, the call becomes a normally dialed call and the originator will receive the appropriate indication on their display.

Night Service

When the system is in Night Service, DID calls forwarded by Call Forward Busy are routed to the specified night number. If the night telephone is busy, subsequent calls receive busy tone.

Recorded Announcement for Call Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) is activated if the call is forwarded to an outgoing external CO trunk with the RANX feature active.

Recovery on Misoperation of Attendant Console

Call Forward takes precedence over the Misoperation feature.

Total Redirection Count

Call Forward Busy redirections is limited to the value defined in the Total Redirection Count limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

Trunk Barring

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Add/change a Call Forward Busy Incoming Call Indicator (ICI) on Attendant Consoles.

2 LD 10 – Allow/deny Call Forward Busy on analog (500/2500 type) telephones.

3 LD 11 – Allow/deny Call Forward Busy on Meridian 1 proprietary telephones.

LD 15 – Add/change a Call Forward Busy Incoming Call Indicator (ICI) on Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant Consoles Options

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ICI	xx CFB	Add a Call Forward Busy ICI key; xx = 0-19.

LD 10 – Allow/deny Call Forward Busy on analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(FBD) FBA	(Deny) allow Call Forward Busy.

LD 11 – Allow/deny Call Forward Busy on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(FBD) FBA	(Deny) allow Call Forward Busy.

Feature operation

No specific operating procedures are required to use this feature.

Call Forward by Call Type

Contents

This section contains information on the following topics:

Feature description 637
Operating parameters 639
Feature interactions 639
Feature packaging 644
Feature implementation 644
Feature operation 647

Feature description

Call Forward by Call Type (CFCT) routes calls receiving a no answer or busy signal to separately defined DNs based on the type of incoming call. The two types of incoming calls are internal and external.

An internal call is defined as a station-to-station call, a Direct Inward System Access (DISA) call, or an incoming call over a trunk route class marked as internal. An external call is an incoming call over a trunk route class marked as external. The trunk route data block (LD 16) allows routes to be defined as internal or external for this feature.

Four options are available at the customer level for Call Forward No Answer: Flexible Call Forward No Answer DN (FDN), Attendant Recall (ATT), Call Forward denied for all telephones (NO), and Hunting (HNT). Call Forward by Call Type (CFCT) is enabled only when the FDN and HNT options are chosen.

In LD 15 Call Forward No Answer is defined by FNAT for external non-DID calls and by FNAL for internal calls. FNAD continues to define Call Forward No Answer for Direct Inward Dialing (DID) trunk calls.

CFCT is allowed or denied for each telephone in LD 10 or LD 11 with Class of Service (CFTA/CFTD). If CFCT is allowed (CFTA), the forwarding destination is also defined in LD 10 or LD 11.

Once enabled, CFCT requires no intervention. How the system initiates Call Forward by Call Type is described below.

When a call is presented to a telephone, the telephone is checked for the appropriate Class of Service (Hunting Allowed (HTA), Call Forward No Answer (FNA), Call Forward by Call Type (CFTA)). The incoming call is then checked to determine if it is a telephone, DISA, or trunk call. If it is a trunk call, the trunk route is checked to determine whether the call should be treated as an internal or external call. After these checks, internal calls are forwarded to the FDN or Hunt DN of the telephone. External calls are forwarded to the External Flexible DN (EFD) or External Hunt (EHT) DN of the telephone.

The order in which the system handles no answer and busy calls is an important consideration when implementing this feature. The order of precedence is listed below.

Calls to telephones that do not answer:

- Call Forward All Calls
- Message Waiting
- Call Forward No Answer
- Attendant Recall

Calls to busy telephones:

- Call Forward All Calls
- Hunting
- Call Waiting or Camp-On
- Message Waiting Forward Busy
- Call Forward Busy

Operating parameters

Attendant Administration does not support the entry of the new EFD and EHT Class of Service required for Call Forward by Call Type.

The following trunk routes can be defined as internal or external call types for CFCT: CO, DID, FX, ATVN, CCA, TIE, WATS, and CSA.

Incoming DISA calls are always treated as internal calls irrespective of the trunk route class mark defined for the incoming trunk.

If an incoming call has been modified by Call Forward All Calls or Hunting, the Class of Service and forwarding DN are obtained from the dialed DN. This applies when Call Forward No Answer specified at the customer level is HNT or FDN.

Feature interactions

Attendant

An attendant-extended call is classified internal or external by the originating telephone or class mark of the trunk type. This is the case whether or not the attendant has released before forwarding occurs.

Attendant Alternative Answering

If Call Forward by Call Type is enabled on the Attendant Alternative Answering (AAA) DN, calls are forwarded based on the Call Type of the originator.

Attendant Break-In to Inquiry Calls

The operation of Call Forward by Call Type is overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

Automatic Timed Recall

Calls eligible for Flexible Call Forward No Answer treatment, and handled by Call Forward by Call Type, use the Call Forward No Answer timer in the Customer Data Block as the recall timer for attendant extended calls. Irrespective of the relative timeout for Automatic Timed Recall, the ringing continues as long as allowed by the Call Forward No Answer Timer.

Call Forward All Calls

If a call is unanswered at the forwarded DN, the telephone that has Call Forward All Calls activated is checked for the Class of Service and the call forward DN. If a chain of call forwarding occurs, the Class of Service and the forward DN for Call Forward No Answer are obtained from the first telephone in the chain. This applies when FDN and HNT have been specified for Call Forward No Answer at the customer level.

Call Forward No Answer

The sequence for forwarding unanswered calls is Call Forward All Calls, Message Waiting, Call Forward No Answer, then Attendant Recall (if the call is attendant-extended). The same sequence is used when Call Forward by Call Type is active for the customer.

Call Forward No Answer, Second Level

To implement CFCT for Second Level Call Forward No Answer eligible calls, the originating party's call type is checked. If it is internal, the call is forwarded to the Flexible Call Forward No Answer DN (FDN). If it is external, the call is forwarded to the External Flexible DN (EFD).

Call Forward Save on SYSLOAD

The Call Forward status of each telephone can be saved as part of the data dump routine and reinstated following a SYSLOAD operation.

Call Forward, Break-In and Hunt Internal/External Network Wide

If the Internal/External definition in LD 15 is set to YES, a call is treated as internal or external on a network wide basis.

Call Redirection by Time of Day

Call Forward by Call Type (CFCT) takes precedence over Call Redirection by Time of Day.

If Call Forward by Call Type (CFCT) is enabled with Call Forward No Answer (CFNA) and Call Redirection by Time of Day (CRTOD), unanswered internal calls receiving CFNA are routed to the Flexible CFNA DN, Hunt DN, Alternate Flexible CFNA DN or Alternate Hunt DNs. External calls are routed in the same manner.

If CFNA is enabled with Hunting by Call Type and Call Redirection by Time of Day (CRTOD), unanswered internal calls are redirected to the Hunt DN or Alternate Hunt DN during the alternative time. External calls are routed in the same manner. The alternate time is defined on the called DN's data block.

Calling Party Name Display Denied

During a Call Forward or Call Transfer, the calling party digits and forwarding/transferring party digits are displayed on the terminating set. This display is allowed or denied depending on the Class of Service of the calling set and the forwarding/transferring set. The name of the forwarding/ transferring set is not displayed on the calling and terminating set.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Call Transfer
Network Call Transfer

Calls modified by Call Transfer and Network Call Transfer receive CFCT treatment. If party A (telephone or trunk) calls party B, and B transfers to party C, the forwarding DN and Class of Service are obtained from party C.

Call Waiting Redirection

If Call Forward and Hunt by Call Type (CFCT) is enabled with Call Forward No Answer and Call Waiting Redirection, "no answer" internal calls receiving Call Waiting treatment are routed for CFNA treatment to the Flexible CFNA DN (FDN) or Hunt DN, and "no answer" external calls are routed for CFNA treatment to the External Flexible CFNA DN (EFD) or External Hunt DN (EHT).

Conference

Calls modified by Conference receive CFCT treatment for the conferenced telephone. If party A calls party B, and B tries to conference in party C, the forwarding DN and Class of Service are that of C. For example, Joan and Bob are in conversation, and they try to conference in Mack. Mack is not at his desk, so the attempted conference call is sent to the destination associated with Mack's telephone.

Direct Inward Dialing (DID)

Eligibility of a DID call for Call Forward by Call Type is determined by allowing or denying the type of call in the Customer Data Block (FNAD prompt). The decision to treat a DID call as internal or external is made on a trunk route basis.

Group Hunting Queuing Limitation Enhancement

An external call is made to the PLDN. An idle group hunt list member station is rung but does not answer. If the member station has Call Forward No Answer (FNA) or Call Forward by Call Type Allowed (CFTA) Class of Service, then the call is transferred to the attendant after the number of ring cycles defined for Call Forward No Answer has been reached. If the call is an internal call, then the system searches for another idle group hunt list member.

ISDN Semi Permanent Connections for Australia

ISPC calls are handled according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Message Center

Message Center uses the Flexible Call Forward No Answer DN (FDN) of the called telephone to route no answer calls. If CFCT is enabled, unanswered internal calls use the FDN to route a call. Unanswered external calls use the External Flexible DN (EFD) to route a call.

Multiple Appearance Directory Numbers

Call redirection parameters like Call Forward No Answer are derived from the TN data block of the prime appearance of the called Multiple Appearance Directory Number. If there is more than one prime appearance, the parameters are selected from the last TN in the DN block.

If more than one prime appearance of a MADN exists, the following information must be considered prior to configuring call redirection parameters for MADNs.

The DN Block organizes MADN information in numerical TN order. The TN with the highest numerical value (000-0-06-03) is placed at the beginning of the list. The list then continues in descending order with the lowest numerical TN (000-0-03-01) at the end of the list. Service change activity affects the organization of the DN list as described in the following paragraphs.

- If a telephone undergoes Service Change, its TN is moved to the beginning of the DN list, irrespective of the numerical value. This telephone remains at the beginning of the list until another service change or a SYSLOAD.

- If a DN appears on analog (500/2500 type) telephones, and Meridian 1 proprietary telephones, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the list. Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A Service Change to an analog (500/2500 type) telephone moves its TN to the beginning of the list. A Service Change to a Meridian 1 proprietary telephone moves its TN to the end of the list.
- A SYSLOAD restructures the list back to numerical TN order, with analog (500/2500 type) telephones at the top and Meridian 1 proprietary telephones at the bottom. Call redirection parameters continue to be derived as described in the preceding paragraphs.

Recovery on Misoperation of Attendant Console

Call Forward takes precedence over the Misoperation feature.

Trunk Barring

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

Feature packaging

Call Forward by Call Type is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Call Forward by Call Type for a customer.

2 LD 16 – Define a trunk route as internal or external for Call Forward by Call Type.

3 LD 10 – Enable Call Forward by Call Type for analog (500/2500 type) telephones.

4 LD 11 – Enable Call Forward by Call Type for Meridian 1 proprietary telephones.

LD 15 – Enable Call Forward by Call Type for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Call Redirection
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- FNAD	(HNT) ATT NO FDN	Treatment for incoming DID calls.
- FNAT	(HNT) ATT NO FDN	Treatment for incoming external calls.
- FNAL	(HNT) ATT NO FDN	Treatment for incoming internal calls.
- CFNA	1-(4)-15	Number of ringing cycles for CFNA.

LD 16 – Define a trunk route as internal or external for Call Forward by Call Type.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
RCLS	(EXT) INT	Route class marked as (external) or internal.

LD 10 – Enable Call Forward by Call Type for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
HUNT	xxxx	Hunt DN for internal calls.
CLS	(CFTD) CFTA	(Deny) allow Call Forward by Call Type Telephone. Must have Hunting (HTA) and Call Forward No Answer (FNA) allowed.
FTR	EFD xxxx EHT xxxx FDN xxxx	Flexible Call Forward No Answer DN for external calls. Hunt DN for external calls. Flexible Call Forward No Answer DN for internal calls.

LD 11 – Enable Call Forward by Call Type for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FDN	xxxx	Flexible Call Forward No Answer DN for internal calls.
CLS	(CFTD) CFTA	(Deny) allow Call Forward by Call Type Telephone. Must have Hunting (HTA) and Call Forward No Answer (FNA) allowed.
EFD	xxxx	Flexible Call Forward No Answer DN for external calls.

HUNT	xxxx 000	Hunt DN for internal calls. Short Hunt for internal calls.
EHT	xxxx 000	Hunt DN for external calls. Short Hunt for external calls.
LHK	xx	Last hunt key number for internal and external calls

Feature operation

No specific operating procedures are required to use this feature.

Call Forward Destination Deactivation

Contents

This section contains information on the following topics:

Feature description 649
Operating parameters 650
Feature interactions 652
Feature packaging 652
Feature implementation 653
Feature operation 655

Feature description

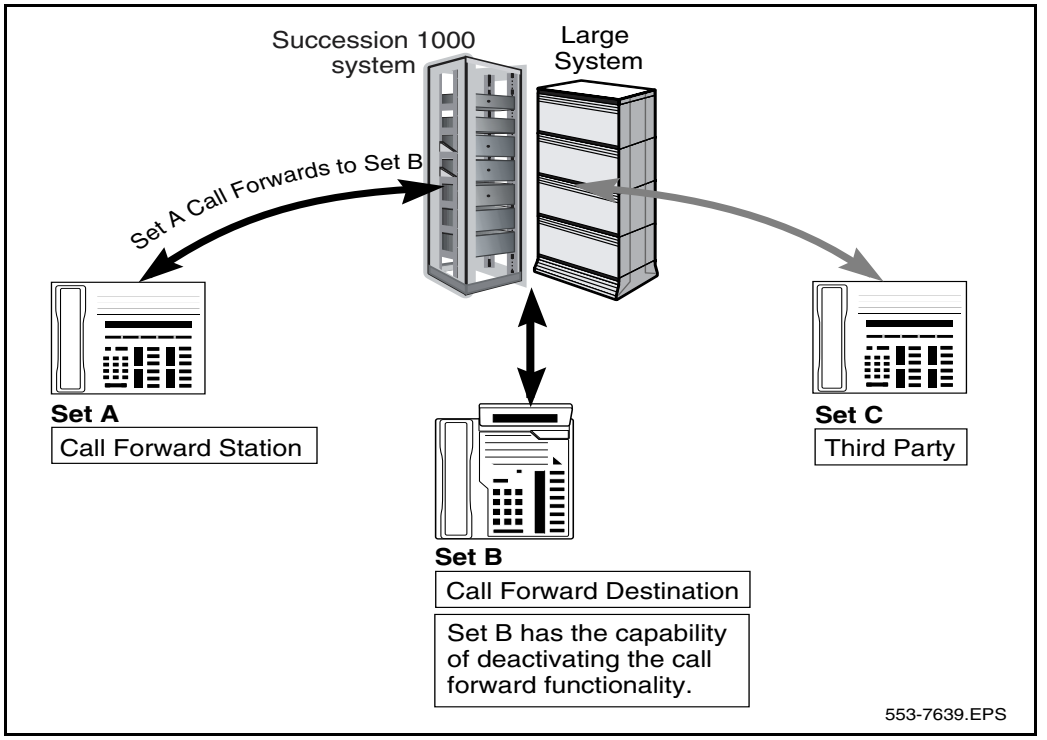
The existing Call Forward All Calls feature allows users to divert incoming calls from a telephone set. The activation/deactivation of Call Forward All Calls must occur from the originating telephone. The Flexible Feature Codes and Remote Call Forward features allowed users the extended flexibility to activate/deactivate Call Forward All Calls from within the system, or outside the local network using Direct Inward System Access.

Call Forward Destination Deactivation (CFDD) permits the call forwarded destination to deactivate the Call Forward All Calls functionality on the call forward station. As illustrated in Figure 10, if set A is call forwarded to set B, then Set B can deactivate call forward.

This feature also allows a user to deactivate call forward by using Remote Call Forward deactivate Flexible Feature Code.

In China, the Call Forward Destination Deactivation feature requires an octothorpe (#) as an end of dial delimiter when entering FFC's to deactivate.

Figure 10
Call Forward Destination Deactivation Capability



Operating parameters

The feature is applicable to all systems.

Call Forward Destination Deactivation is only supported on Call Forward. This feature is not supported on Internal Call Forward.

The existing Call Forward All Calls features allows users to program a call forward station within a system switch or Public Switching Telephone Network. The Call Forward Destination Deactivation feature is designed for stand alone application only. Therefore, both the destination and originator must belong to the same customer on the system.

The call forwarded destination can deactivate the Call Forward All Calls functionality. However, the call forwarded destination cannot activate the call forward functionality from the originating set unless using the RCFA FFC.

CFDD can be activated on proprietary, ISDN Basic Rate Interface and Analog (500/2500 type) sets by dialing the CFDD FFC. 16-button Dual-Tone Multi-Frequency sets can use one of the A,B,C or D function keys, configured as CFDD, or they can also dial CFDD FFC to use this feature.

CFDD can be activated on Meridian proprietary, ISDN Basic Rae interface and Analog (500/2500 type) sets by dialing the CFDD FFC. An analog 500-type set with a dial pulse Class of Service cannot dial an octothorpe (#) as the end of dial delimiter. To Activate CFDD, the call forwarded destination of an analog 500-type set has to dial the string of digits (as defined in LD 15) for the end of dial delimiter.

An analog 500-type set with a dial pulse Class of Service cannot dial an octothorpe (#) as the end of dial delimiter. To activate CFDD on an analog 500-type set, the dial string digits (the STRG prompt in LD 15) must be configured for the end of dial delimiter.

If the string to indicate end of dialing (STRG prompt in LD 15) is defined, then analog (500/2500 type), 16-button DTMF, ISDN BRI and proprietary sets must dial string digits as an end of dial delimiter to activate CFDD.

In China, to activate CFDD a user must dial an octothorpe (#) as an end of dial delimiter. In this market, an analog 500-type telephone does not support this feature.

An Automatic Call Distribution (ACD) agent can only activate this feature from their personal Directory Number key. This feature cannot be activated on the ACD in calls key.

If the call forward station has a Prime DN and Secondary DN defined, then the Call Forward Destination Deactivation feature only considers the Prime DN to forward a call. Therefore, during the feature operation, the call forwarded destination's dialed DN of Set A is compared with the call forward station's Prime DN.

Feature interactions

Call Forward All Calls

If a user's call forwarded Directory Number (DN) is defined as DN of Set B and set A dials the CFW FFC to activate call forward, then Set A gets forwarded to Set B. Set B can deactivate CFW on set A by dialing the CFDD FFC.

Call Forward, Remote

Remote Call Forward (RCFW) and Call Forward Destination Deactivation (CFDD) provide the same functionality but are activated differently. CFDD does not require the call forward station's control password to deactivate the call forward functionality on the call forward station.

The call forwarded destination can use the Remote Call Forward deactivation FFC as well as CFDD to deactivate the Call Forward All Calls functionality on the call forward station.

Meridian Mail

Meridian Mail cannot deactivate the CFW functionality on the call forward station by using CFDD FFC.

Feature packaging

Call Forward Destination Deactivation (CFDD) requires Flexible Feature Code (FFC)/China Flexible Feature Code (CHFFC) package 139.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure Dial String in Customer Data Block.

2 LD 57 – Configure Flexible Feature Codes for Call Forward Destination Deactivation.

3 LD 18 – Configure Call Forward Destination Deactivation FFC on 16-button DTMF Set.

The Call Forward All Calls feature is configured in LD 10 for Analog (500/2500 type) telephones and LD 11 for proprietary telephones.

LD 15 – Configure Dial String in Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FFC	Flexible Feature Code gate opener.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- FFCS	YES	Change end of dialing digits.
- - STRL	1-3	Number of digits to indicate end of dialing.
- - STRG	xxx	String to indicate end of dialing. Outpulsing of Asterisk (*) and Octothorpe (#) (OPAO) package 104 is required to outpulse (*) and (#) in the string.

LD 57 – Configure Flexible Feature Codes for Call Forward Destination Deactivation.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Flexible Feature Confirmation Tone. YES = confirmation tone is required.
CODE	DEAF	Deactivate Feature.
- DEAF	xxxx	Enter Flexible Feature Code on a set.
CODE	CFWA	Call Forward All Calls Activate Code.
- CFWA	xxxx	Enter Flexible Feature Code to activate.
CODE	CFWD	Call Forward All Calls Deactivation Code.
- CFWD	xxxx	Enter Flexible Feature Code to deactivate.
CODE	CFWV	Call Forward All Calls Verify Code.
- CFWV	xxxx	Enter Flexible Feature Code to verify.
CODE	CFDD	Call Forward All Calls Destination Deactivation Code.
- CFDD	xxxx	Enter Flexible Feature Code.

LD 18 – Configure Call Forward Destination Deactivation FFC on 16-button DTMF Set.

Prompt	Response	Description
REQ	NEW CHG	Add, or Change 16-button data block.
TYPE	ABCD	16-button data block.
TBNO	1-254	Number of ABCD Table to be added or changed.
DFLT	1-254	Number of default function table.
PRED	YES	Function table is changed for predial. NO = default mnemonics are used.
- A	CFDD	Call Forward Destination Deactivation FFC assignment of key.
- B	CFWA	Call Forward All Call Activation FFC assignment of key.
...		
Note: Call Forward Destination Deactivation FFC function can be assigned to any A, B, C or D key of the 16-button DTMF set.		

ISDN BRI sets can deactivate the Call Forward All Call feature. To set up the digital subscriber loop and terminal service profile for a BRI set refer to Overlay 27.

Feature operation

To enable the Call Forward Destination Deactivation feature, complete the following:

1. Go off-hook on the “call forward to” Directory Number of the call forwarded set and listen for dial tone.
2. Dial the CFDD FFC followed by DN of the call forward originator and end of dial delimiter.
3. Response provided.

The following responses are provided to the user of the call forwarded destination.

1 If originator's call forward DN is the same as active DN of destination then the call forward feature on originator is deactivated. If the confirmation tone in LD 57 is enabled, then a confirmation tone or speech is provided to the destination set. Otherwise, silence is provided.

2 If the originator's call forward DN does not match the active DN of the destination, then an overflow tone is provided to the destination set.

3 If the Call Forward All Calls functionality on the originator set is already deactivated, then an overflow tone is provided to the destination set.

4 If the originator does not have call forward all calls defined, then an overflow tone is provided to the destination set.

16-Button DTMF set

To activate CFDD on this set, the user must press one of the ABCD function keys that is defined as CFDD FFC, followed by DN of call forward station set and end of dial delimiter.

Deactivating Multiple Appearance DN

The Call Forward Destination Deactivation feature operation remains the same. However, if the call forward station is a Multiple Appearance DN (MADN) set, then the station's Prime DN is considered to find the Call Forward All Calls functionality for deactivation.

If more than one MADN call forward stations have Call Forward All Calls defined and the call forward DN on one or more MADN call forward stations matches the active DN of Set B, then the call forward functionality on one or more MADN call forward stations is deactivated.

Any one of MADN call forwarded destination station is allowed to deactivate the call forward function on the call forward station by using the CFDD FFC.

Call Forward External Deny

Contents

This section contains information on the following topics:

Feature description 657
Operating parameters 658
Feature interactions 659
Feature packaging 660
Feature implementation 660
Feature operation 661

Feature description

This enhancement provides the option to restrict, on a per-telephone basis, the DN that can be programmed for Call Forward All Calls to internal DNs only. Internal DNs are defined as:

- DNs that terminate on an analog (500/2500 type) telephone
- DNs that terminate on a Meridian 1 proprietary telephone
- DNs that terminate on a data terminal defined in LD 10 or LD 11
- Attendant DNs or Centralized Attendant Service (CAS) local attendant DNs
- Listed DNs (LDNs)
- Message Center DNs as defined in LD 23

External DNs include (but are not limited to) trunk access codes, Coordinated Dialing Plan (CDP) steering codes, Basic and Network Alternate Route Selection (BARS/NARS) access codes, Electronic Switched Network (ESN) Location Codes, non-message center Automatic Call Distribution (ACD) numbers, Call Park numbers, and Direct Inward Services Access numbers.

When Call Forward External Deny is enabled for a telephone:

- A user trying to forward calls from an analog (500/2500 type) telephone to an external DN receives overflow tone. The telephone is not call forwarded.
- A user trying to forward calls from a Meridian 1 proprietary telephone to an external DN receives overflow tone and the lamp associated with the Call Forward key of the telephone flashes. The telephone is not call forwarded.
- A user trying to forward calls from a Meridian digital or a display telephone to an external DN receives overflow tone. The telephone is not call forwarded and one of the following messages is displayed:
  - **Release and try again** (M2317 telephones)
  - **Release, check, and try again** (M3000 telephones)
- A user trying to forward calls from a data module to an external DN does not receive overflow tone. Calls are not forwarded and one of the following messages is displayed:
  - **Invalid data forward number** (M2317 telephones)
  - **Data calls not forwarded** (M3000 telephones)

Operating parameters

External DNs cannot be used with Call Forward All Calls if Call Forward External Deny is enabled for the telephone.

Both ESN access codes and CDP steering codes are considered external DNs, and cannot be used as a Call Forward All Calls DN if Call Forward External is denied for the telephone.

The number of digits specified in LD 10 or LD 11 for the Call Forward DN must be equal to or greater than the number of digits of any other internal DN.

Attendant Administration cannot change Call Forward External Deny Class of Service.

Feature interactions

Automatic Call Distribution (ACD)

If Call Forward External Deny (CFXD) is enabled, Call Forward to an ACD DN is allowed only if the ACD DN is a message center.

Call Forward All Calls

This feature overrides other Call Forward All Calls parameters. For example, if Call Forward to Trunk Access Code (CFTA) is allowed for the customer, but Call Forward External Deny (CFXD) is enabled for the telephone, CFXD takes precedence and call forwarding to a trunk access code is denied.

Calling Party Privacy

If an incoming ISDN trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the outgoing trunk route on the tandem node also has CCP provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (that is, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Network Call Forward

Call Forward External Deny restricts a telephone from being forwarded unconditionally to a number that is not on the home switch. Therefore, Call Forward External Deny and the Integrated Services Digital Network Primary Rate Interface (ISDN PRI) feature Network Call Forward are mutually exclusive.

Recovery on Misoperation of Attendant Console

Call Forward takes precedence over the Misoperation feature.

Trunk Barring

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Allow/deny Call Forward External Deny for analog (500/2500 type) telephones.

2 LD 11 – Allow/deny Call Forward External Deny for Meridian 1 proprietary telephones.

LD 10 – Allow/deny Call Forward External Deny for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	CFXA	Allow Call Forward to an external DN CFXD = Deny Call Forward to an external DN (default).

LD 11 – Allow/deny Call Forward External Deny for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	CFXA	Allow Call Forward to an external DN. CFXD = Deny Call Forward to an external DN (default).

Feature operation

No specific operating procedures are required to use this feature.

Call Forward No Answer, Second Level

Contents

This section contains information on the following topics:

Feature description 663
Operating parameters 665
Feature interactions 666
Feature packaging 671
Feature implementation 671
Feature operation 672

Feature description

Second Level Call Forward No Answer enhances Flexible Call Forward No Answer by forwarding unanswered calls twice. The following example best illustrates this enhancement.

Party A places a call to extension 5000, which does not answer. Extension 5000 has Call Forward No Answer (CFNA) allowed and extension 6000 defined as its CFNA number. The call forwards to extension 6000. This is the first level CFNA.

Extension 6000 also does not answer the call. This telephone has a Call Forward No Answer and Second Level Call Forward No Answer allowed Class of Service (FNA and SFA). As it has a CFNA number of 7000, it forwards there. This is the second level of Call Forward No Answer. Note that the forwarding DN is always obtained from the currently ringing telephone.

If extension 7000 does not answer the call, one of two things may occur:

- If the original call is a Direct Inward Dialing (DID) or internal call, the forwarded call continues to ring until answered or the calling party disconnects.
- If the original call is extended by the Attendant Console, Attendant Recall occurs.

Second Level Call Forward No Answer uses the same customer-level timer as Flexible Call Forward No Answer to determine the number of rings before forwarding a call.

Telephones with a message waiting allowed (MWA) Class of Service should have the Message Center DN defined as their FDN. Calls to these telephones forward to the Message Center and are not eligible for Second Level Call Forward No Answer.

Call Forward No Answer Second Level for Message Waiting Allowed Telephones, enables an SFA Class of Service to be defined on telephones with a Message Waiting Allowed (MWA) Class of Service. Thus, a message waiting indication can be activated at the originally dialed DN for Second Level CFNA calls terminating at a message center.

Requirements at the dialed DN for first-level CFNA are as follows:

- Flexible Call Forward No Answer or Hunting is allowed at the customer level for the incoming call type (DID, non-DID, or internal).
- The telephone has an FNA Class of Service.
- The terminating call has rung for the number of rings specified for CFNA or DFNA in the Customer Data Block (LD 15).
- The forwarding DN (Flexible Call Forward No Answer DN [FDN], Coordinated Dialing Plan DN [CDP DN], External Flexible DN [EFD], Hunting [HNT], or External Hunt [EHT]) must be distinct from the ringing DN and be a valid number within the system.

Requirements at the originally called telephone DN for Second Level Call Forward No Answer are as follows:

- Flexible Call Forward No Answer or Hunting is allowed at the customer level for the incoming call type (DID, non-DID, or internal).
- The telephone has SFA and FNA Class of Service.
- Call Forward No Answer has occurred only once prior to ringing this telephone.
- The forwarding DN (FDN, EFD, Hunt, or EHT) must be distinct from the ringing DN and must be a valid number within the system.

The order of precedence for activation of first level Call Forward No Answer is as follows:

- Call Forward All Calls
- Message Waiting
- Call Forward No Answer, and
- Attendant Recall.

The order of precedence for activation of Second Level Call Forward No Answer is as follows:

- Call Forward All Calls
- Second Call Forward No Answer (CFNA calls only)
- Attendant Recall

Operating parameters

A maximum of two levels of Call Forward No Answer are allowed for an unanswered call.

Calls directed to an attendant or Automatic Call Distribution (ACD) Message Center cannot have Second Level Call Forward No Answer.

Attendant Administration cannot change the SFA/SFD Class of Service. Error messages are generated if changes made to the Forward No Answer or Hunt Class of Service conflict with the SFA/SFD Class of Service.

Feature interactions

Automatic Timed Reminders

When Call Forward No Answer is activated on a telephone, the slow answer timer begins only after the call reaches its final destination.

Call Detail Recording on Redirected Incoming Calls

The Call Detail Recording on Redirected Incoming Calls feature does not affect how these features operate; however, it does provide information about the answering party in the CDR ID field if incoming calls have been redirected by any one of these features.

Call Forward All Calls

Second Level Call Forward No Answer uses the final (ringing) telephone in the chain to obtain Class of Service and forwarding DN information.

First level treatment operates in the following manner. Suppose that Party A calls Party B, and Party B has programmed Call Forward All Calls to Party C. Flexible Call Forward No Answer will forward a No Answer call at Party C to the forwarding directory number associated with Party B, the dialed DN.

Call Forward by Call Type

To implement Call Forward by Call Type for Second Level Call Forward No Answer eligible calls, the originating party's call type is checked. If it is internal, the call is forwarded to the Flexible Call Forward No Answer DN (FDN). If it is external, the call is forwarded to the External Flexible DN (EFD).

Call Forward No Answer

Second Level Call Forward No Answer applies to the Hunt and Flexible Call Forward No Answer options. This is implemented by defining the FNAD, FNAT, or FNAL prompts in LD 15 as FDN or HNT. If the attendant option is defined, an unanswered call goes to the attendant queue and is not eligible for Second Level Call Forward No Answer.

Call Redirection by Time of Day

Existing Second Level CFNA allows unanswered calls to receive Call Forward No Answer treatment twice. Call Redirection by Time of Day (CRTOD) parameters are obtained from the last rung Directory Number. A maximum of two levels of CFNA is allowed for an unanswered call.

Call Waiting Redirection

The existing Second Level CFNA treatment is applicable to Call Waiting calls redirected by CFNA (first level) with the Call Waiting Redirection feature which are still not answered at the last rung DN.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Directory Number Delayed Ringing

The Directory Number Delayed Ringer (DNDR) feature allows the SCN/MCN (non ringing keys) to actually ring after a definable period of time (DNDR prompt in LD 11). If the time before CFNA takes effect is less than the DNDR time for a particular set, CFNA will forward this call before any SCN/MCN keys can ring on this set. Note that CFNA is defined in the number of rings and DNDR is defined in seconds.

If the Forward DN set is busy or invalid when the call is forwarded, the call will return to the originally called set. However, the DNDR delay timer will be reapplied to the called set if DNDR is defined.

If a call is forwarded, as per existing operation, this call will be treated as a new incoming call to the forward DN. For example, if the forward DN has a DNDR value defined, a new timer will begin timing according to the forward DN's DNDR delay.

Distinctive/New Distinctive Ringing

The ringing cadence for all telephones in a chain of call redirections remains the same as for the original DN called.

Flexible Call Forward No Answer

If Second Level Call Forward No Answer is disabled, Flexible Call Forward No Answer operates as described.

Group Hunt

Second Level Call Forward No Answer will not be applied to calls that are Group Hunting.

Hunting

A forwarded call may be modified by Hunting if the Call Forward No Answer DN is busy. This call is eligible for Second Level Call Forward No Answer if the SFA Class of Service is allowed and a Call Forward No Answer DN has been defined for the last rung DN.

If Group Hunting is active, Second Level CFNA is not applied.

Message Centers

There are three types of Message Centers:

- Automatic Call Distribution (ACD)
  Calls forwarded to an ACD Message Center are queued, so that no CFNA timeout occurs.
- Attendant
  Calls forwarded to an Attendant Message Center are queued, so no CFNA timeout occurs.
- DN
  An indirect call forwarded to a DN Message Center and not answered by an operator is forwarded to a second level if SFA for DN-MC.

Note: It is recommended that DN Message Center stations be denied CFNA, Call Forward Busy (CFB), Call Forwarding (CFW), and Call Hunting (HUNT).

Multiple Appearance Directory Numbers

Call redirection parameters like Hunt and Call Forward No Answer are derived from the TN data block (LD 20 TNB) of the prime appearance of the called Multiple Appearance Directory Number (MADN). If there is more than one prime appearance, the parameters are selected from the last TN in the DN block for the DN (LD 22 DNB).

If more than one prime appearance of a MADN exists, the following information must be considered prior to configuring call redirection parameters for MADNs.

The DN Block organizes MADN information in numerical TN order. The TN with the highest numerical value (000-0-06-03) is placed at the beginning of the list. The list then continues in descending order with the lowest numerical TN (000-0-03-01) at the end of the list. Service change activity affects the organization of the DN list as described in the following paragraphs:

- If a telephone undergoes Service Change, its TN is moved to the beginning of the DN list, irrespective of the numerical value. This telephone remains at the beginning of the list until another Service Change or a SYSLOAD.
- If a DN is assigned as a Prime DN on a telephone and as a secondary DN on one or more telephones, the DN list is still organized as described in the preceding paragraphs. If only one prime appearance of a DN exists, however, call redirection parameters are derived from the TN of the prime appearance telephone, even though it may not be at the end of the list. A prime appearance is always the first TN used when the system looks for call redirection instructions.

- If a DN appears on analog (500/2500 type) telephones, and Meridian 1 proprietary telephones, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the list. Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the beginning of the list. A Service Change to a Meridian 1 proprietary telephone moves its TN to the end of the list.
- A SYSLOAD restructures the list with analog (500/2500 type) telephones at the top and Meridian 1 proprietary telephones at the bottom. Call redirection parameters continue to be derived as described in the preceding paragraphs.

Recovery on Misoperation of Attendant Console

Call Forward takes precedence over the Misoperation feature.

Slow Answer Recall

When a Call Forward No Answer call is unanswered at a telephone eligible for Second Level Call Forward No Answer, and the call was extended by an attendant, Second Level Call Forward No Answer takes precedence over Slow Answer Recall. If the telephone has a Second Level Call Forward No Answer Denied Class of Service, the system performs Slow Answer Recall for the unanswered call.

Total Redirection Count

If a call has attempted Call Forward No Answer and was extended by the attendant, the call will not be intercepted when the Total Redirection Count limit has been exceeded. The call will continue to ring the telephone until recalled to the attendant.

If Overflow (OVF), Busy (BSY), or Source (SRCx) is configured as Intercept Treatments, a call attempting Call Forward No Answer, that exceeds the Total Redirection Count limit, will not be intercepted. Further redirections are prohibited and the call continues to ring the current telephone.

Trunk Barring

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Assign Message Center to allow the Message Waiting indication.
2. LD 10 – Add/change Second Level Call Forward No Answer for analog (500/2500 type) telephones.
3. LD 11 – Add/change Second Level Call Forward No Answer for Meridian 1 proprietary telephones.

LD 15 – Assign Message Center to allow the Message Waiting indication.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(MCX) MCI	(Exclude) include Message Center.

LD 10 – Add/change Second Level Call Forward No Answer for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(FND) FNA (MWD) MWA (SFD) SFA	(Deny) allow Forward No Answer. (Deny) allow Message Waiting. (Deny) allow second level CFNA SFA can be implemented with an MWA Class of Service.
FTR	FDN xxxx...x	Flexible Call Forward No Answer DN.

LD 11 – Add/change Second Level Call Forward No Answer for Meridian 1 proprietary telephones.

Prompt	**Response**	**Description**
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FDN	xxx...x	Flexible Call Forward No Answer DN.
CLS	(FND) FNA (MWD) MWA (SFD) SFA	(Deny) allow Forward No Answer. (Deny) allow Message Waiting. (Deny) allow) Second Level CFNA SFA can be implemented with an MWA Class of Service.

Feature operation

No specific operating procedures are required to use this feature.

Call Forward No Answer/Flexible Call Forward No Answer

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Feature description

Call Forward No Answer automatically forwards unanswered calls to another DN. The customer can specify the number of rings (1 to 15) before the system invokes Call Forward No Answer. The default is four rings.

Four options are available at the customer level for Call Forward No Answer:

- deny for all telephones
- route all unanswered calls to the attendant
- route all unanswered calls to the Hunt number defined for the telephone
- route all unanswered calls to the Flexible Call Forward No Answer DN defined for the telephone

Flexible Call Forward No Answer allows the customer to specify, on a per-telephone basis, where an unanswered call should be routed. This is independent of the Hunt number assigned to the telephone. This capability is implemented on a per-customer basis and can be specified for Direct Inward Dialing (DID) and non-DID call types. When activated, a call to a telephone that does not answer within the specified number of ring cycles is forwarded to the Flexible Call Forward No Answer DN (FDN) associated with that telephone.

The Flexible Call Forward No Answer operation applies only to individual DN calls of analog (500/2500 type) telephones, and not to Automatic Call Distribution (ACD) calls.

A call is forwarded under the following conditions:

- The Class of Service of the dialed telephone is Forward No Answer allowed.
- Flexible Call Forward No Answer is enabled at the customer level.
- The call has rung the specified number of times.
- The Call Forward No Answer DN (FDN) is valid and has been assigned.

System or telephone features such as Hunting and Call Forward All Calls may result in the presentation of a call to a telephone that is different from the dialed DN. In these cases, if the call is eligible for Flexible Call Forward No Answer, it is forwarded to the DN specified for the dialed DN, not the ringing DN.

When you use Multiple Appearance DNs (MADNs), call redirection is determined based on the Terminal Number (TN) order in your DN block. To determine the TN order, print the DN block from LD 20 or LD 22 (TYPE = DNB). When a call comes in to a MADN, the system begins a search to determine how the call will be handled. Using the TN list you printed, the system performs the following search, beginning at the bottom of the TN list and working up.

1. Search for the first Prime DN appearance of the MADN with Call Forward All Calls activated.
2. If there are no Prime DN appearances, the TN at the bottom of the list controls call redirection.

   Note: The search does not necessarily determine the highest or lowest numerical TN.

Operating parameters

Calls are forwarded one step only. For Call Forward No Answer enhancements, refer to the Call Forward, Second Level module.

Incoming calls on private lines with the Restricted Call Modification option enabled are not forwarded.

Flexible Call Forward No Answer DN (FDN) can be assigned to telephones with Message Waiting Allowed Class of Service. This is irrespective of the telephone's Class of Service and how forward no answer is specified in the Customer Data Block. Message Center always uses the FDN associated with the telephone to route unanswered calls.

Feature interactions

Advice of Charge for EuroISDN

Calls charged with Advice of Charge that are either transferred, extended or redirected to another set using Call Forward No Answer are charged against the last set that answers the call and the controlling set releases.

Attendant Administration

Attendant Administration can assign and change a Flexible Call Forward No Answer DN with the function key on the Attendant Console.

Attendant Alternative Answering

When Attendant Alternative Answering (AAA) DN does not answer, the call can be forwarded by CFNA to the DN defined as the CFNA DN for the originally dialed DN. If the originally dialed DN is the attendant, the call is forwarded to the CFNA DN defined for the AAA DN.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the Call Forward No Answer feature. If the blocked DN of the set has the Call Forward No Answer feature active when the SACP key is pressed to ring the DN, the DN will ring until answered or disconnected. No Call Forward No Answer will be done for the Attendant Blocking of DN call.

Attendant Break-In to Inquiry Calls

The operation of Call Forward No Answer is overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Call Forward No Answer that may be applied to the set.

Attendant Overflow Position

A call rerouted through Attendant Overflow Position will Call Forward to the forwarding DN only if it is the Prime DN or a single appearance DN on that telephone.

Automatic Call Distribution

The Flexible Call Forward No Answer operation does not apply to Automatic Call Distribution (ACD) calls.

Automatic Set Relocation

Calls will not forward no answer to a telephone that is being relocated

Automatic Timed Recall

Flexible Call Forward No Answer timing takes precedence over Automatic Timed Recall timing. Irrespective of the relative time-out intervals for each feature, ringing continues as long as allowed by Call Forward No Answer.

Automatic Timed Reminders

When Call Forward No Answer is activated on a telephone, the slow answer timer begins only after the call reaches its final destination.

Call Detail Recording on Redirected Incoming Calls

The Call Detail Recording on Redirected Incoming Calls feature does not affect how the Call Forward No Answer feature operates; however, it does provide information about the answering party in the CDR ID field if incoming calls have been redirected by any one of these features.

Call Forward All Calls

Suppose that party A calls party B, and party B has programmed Call Forward All Calls to party C. Flexible Call Forward No Answer will forward a No Answer call at party C to the FDN associated with party B, the dialed DN.

Call Forward by Call Type

The sequence for forwarding unanswered calls is Call Forward All Calls, Message Waiting, Call Forward No Answer, then Attendant Recall (if the call is attendant-extended). The same sequence is used when Call Forward by Call Type is active for the customer.

Call Forward/Hunt Override Via Flexible Feature Code

Call Forward No Answer is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Call Forward No Answer, Second Level

Second Level Call Forward No Answer applies to the Hunt and Flexible Call Forward No Answer options. This is implemented by defining the FNAD, FNAT, or FNAL prompts in LD 15 as FDN or HNT. If the attendant option is defined, an unanswered call goes to the attendant queue and is not eligible for Second Level Call Forward No Answer.

Call Page Network Wide

Call Page Network Wide (PAGENET) does not block a station set from being programmed to Call Forward No Answer to an external Paging trunk. At call termination time, calls that are forwarded to an external PAGENET uncontrolled trunk are not blocked. However, calls forwarded to an external PAGENET controlled trunk are given access denied intercept treatment at the Paging node.

Call Redirection by Time of Day

Call redirection parameters for Call Forward No Answer are obtained from the originally dialed Directory Number. When Call Redirection by Time of Day (CRTOD) is activated, unanswered calls given CRTOD treatment are forwarded with CFNA according to the time of day. No changes are made to the existing CFNA feature.

Call Waiting

If a call to a telephone gets CFNA treatment to another telephone that is busy, Call Waiting and Camp-On do not apply. The call will attempt to terminate on the original DN again.

Call Waiting Redirection

Per existing Call Forward No Answer feature operation, the call redirection parameters for CFNA are obtained from the originally dialed DN for redirected calls.

Existing Network CFNA treatment is applied to calls receiving Call Waiting treatment on sets with CFNA and the Call Waiting Redirection feature enabled if the Call Waiting call is not answered before the expiration of the CFNA timer and the CFNA DN is on another node.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Camp-On

When the Call Forward No Answer timer expires for a ringing camped-on call, the call is given Attendant Recall treatment instead of Call Forward No Answer treatment.

China – Attendant Monitor

If an attendant attempts to monitor a DN which is Call Forwarded and is idle, idle DN treatment is given.

China – Toll Call Loss Plan

Toll pad switching is also provided after call forwarding has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Dial Access to Group Calls

Call Forward No Answer cannot be applied to a Group Call.

Departmental Listed Directory Number

Call Forward No Answer to the attendant operates like Call Forward to 0, and are routed to any idle Attendant Console in the customer group.

Directory Number Delayed Ringing

The Directory Number Delayed Ringing (DNDR) feature allows the SCN/MCN (non ringing keys) to actually ring after a definable period of time (DNDR prompt in LD 11). If the time before CFNA takes effect is less than the DNDR time for a particular set, CFNA will forward this call before any SCN/MCN keys can ring on this set. Note that CFNA is defined in the number of rings and DNDR is defined in seconds.

If the Forward DN set is busy or invalid when the call is forwarded, the call will return to the originally called set. However, the DNDR delay timer will be reapplied to the called set if DNDR is defined.

If a call is forwarded, as per existing operation, this call will be treated as a new incoming call to the forward DN. For example, if the forward DN has a DNDR value defined, a new timer will begin timing according to the forward DN's DNDR delay.

Direct Inward Dialing Call Forward No Answer Timer

Call Forward No Answer takes precedence over the Message Center feature.

Electronic Lock Network Wide/Electronic Lock on Private Lines

For Call Forwarding, the COS and NCOS used for the forwarding call can be taken from either the forwarding set or from the forwarded set, depending on the option defined in the Customer Data Block.

For example, set B call forwards all calls to an external trunk. Set A calls set B. If OPT = CFF in LD 15 (Call Forward forwarded to party's COS and NCOS), the COS and NCOS of set B are used for forwarding the call to the trunk. If OPT = CFO (Call Forward originating party's COS and NCOS), the COS and NCOS of set A are used for forwarding the call to the trunk.

Group Call

Group Call cannot be applied to Call Forward No Answer.

Group Hunt

Call Forward No Answer (CFNA) can optionally be configured to use a Pilot DN. This option is available when the HUNT DN or the FDN is defined as a Pilot DN.

If an idle station attempted for termination has CFNA defined, then the station will be rung. If the station does not answer within the customer specified number of ring cycles, then group hunting will continue with the next DN in the group. The calling party will continue to hear ring back tone until one of the termination conditions mentioned in the "Group Hunt" feature description (the last condition is not applicable in this case) is met, or until they releases the call.

Group Hunting Queuing Limitation Enhancement

An external call is made to the PLDN. An idle group hunt list member station is rung but does not answer. If the member station has Call Forward No Answer (FNA) or Call Forward by Call Type Allowed (CFTA) Class of Service, then the call is transferred to the attendant after the number of ring cycles defined for Call Forward No Answer has been reached. If the call is an internal call, then the system searches for another idle group hunt list member.

Hot Line

Any Hot Line telephone can be assigned Call Forward No Answer but it applies only to the two-way Hot Line capability.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an Intercept Computer (ICP) position, a redirected message identification "50" is sent to the ICP computer, when the call is answered.

Incoming Call Indicator Enhancement

When a DID call to a station that is unrestricted from receiving DID calls (UDI Class of Service) is forwarded to a UDI station due to Call Forward No Answer, the call is not RDI-intercepted to the attendant. The dialed party continues to ring. If the call has been forwarded to the attendant, the Call Forward No Answer ICI lights up, and not the RDI-intercept ICI.

Listed Directory Numbers, Network Wide

A Listed Directory Number (LDN) that is assigned to an Incoming Call Indicator (ICI) has a higher priority than a CFNA ICI. When a call is forwarded to an LDN through Flexible DN, the call is presented on an LDN ICI.

Meridian Mail Operator Revert

The Called Party ID can be passed along from the ACD Message Center when Operator Revert is activated. The attendant can now activate the Message Waiting key for the Called Party while active on the redirected call by pressing the Message Indicator key.

For example, Party A calls Party B, which Call Forward No Answers to Meridian Mail. Party A dials 0 and is transferred to a message center with "live" agents. The agent receiving the call sees information for Party B along with the information for Party A, the calling party.

Multi-Party Operations

For Call Transfer with Ring No Answer (RGNA) if the user has selected an option other than Standard, the optional treatment has priority over the CFNA option selected in the LD 15. If the user has chosen the standard option for RGNA, the call will be treated as a normal CFNA call, and handled according to the options selected for CFNA in LD 15. Once the call is routed to a Night DN during recovery of misoperation and the Night DN does not answer, the call is treated according to the NFNA and FDN options chosen for the Night DN. The Night DN can use flexible CFNA DN in two levels. MPO misoperation does not change the operation of the DNFD timer if one has been configured in LD 15.

Multiple Appearance Directory Number Redirection Prime

The MARP TN always controls the call redirection for Call Forward No Answer.

- If a DN is assigned as a Prime DN on a telephone and as a secondary DN on one or more telephones, the DN list is still organized as described in the preceding paragraphs. If only one prime appearance of a DN exists, however, call redirection parameters are derived from the TN of the prime appearance telephone, even though it may not be at the end of the list. A prime appearance is always the first TN used when the system looks for call redirection instructions.
- If a DN appears on analog (500/2500 type) telephones, and Meridian 1 proprietary telephones, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the list. Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the beginning of the list. A service change to a Meridian 1 proprietary telephone moves its TN to the end of the list.
- A SYSLOAD restructures the list back to numerical TN order with analog (500/2500 type) telephones at the top and Meridian 1 proprietary telephones at the bottom. Call redirection parameters continue to be derived as described in the preceding paragraphs.

Network Intercom

Hot Type I calls respect or override all kinds of Call Forward features (Busy, No Answer, All Calls, Internal, etc.) according to per-set definitions. If Call Forward is respected, the call becomes a normally dialed call and the originator will receive the appropriate indication on their display.

Network-Wide Listed Directory Number

A Listed Directory Number (LDN) ICI has a higher priority than a Call Forward No Answer ICI. When a call is forwarded to an LDN through Flexible DN, the call is presented on an LDN ICI.

Night Service enhancements

Any call which has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Periodic Pulse Metering

Metered calls transferred or extended from one station to another using the Call Forward No Answer are charged against the last station at which the call is answered as the controlling station releases. The last party to forward a call onto a metered Periodic Pulse Metering trunk is charged.

Recall to Same Attendant

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call routed to the first available idle attendant.

Recorded Announcement for Calls Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) is activated if the call is forwarded to an outgoing external CO trunk with the RANX feature active.

Recovery on Misoperation of Attendant Console

Call Forward No Answer takes precedence over the Misoperation feature.

Ring Again on No Answer

If an unanswered call is forwarded to another station by Call Forward No Answer, Ring Again on No Answer is applied to the originally dialed station.

Slow Answer Recall for Transferred External Trunks

If the ringing station to which the call has been transferred has Call Forward No Answer active, the call will be transferred to the call forward DN after the specified number of ring cycles.

Total Redirection Count

If a call has attempted Call Forward No Answer and was extended by the attendant, the call will not be intercepted when the Total Redirection Count limit has been exceeded. The call will continue to ring the telephone until recalled to the attendant.

If Overflow (OVF), Busy (BSY), or Source (SRCx) is configured as Intercept Treatments, a call attempting Call Forward No Answer, that exceeds the Total Redirection Count limit, will not be intercepted. Further redirections are prohibited and the call continues to ring the current telephone.

Trunk Barring

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Define Call Forward No Answer for a customer.

2 LD 10 – Add/change Flexible Call Forward No Answer for analog (500/2500 type) telephones.

3 LD 11 – Add/change Flexible Call Forward No Answer for Meridian 1 proprietary telephones.

4 LD 10 – Implement Call Forward No Answer to the Hunt DN on analog (500/2500 type) telephones.

5 LD 11 – Implement Call Forward No Answer to the Hunt DN on Meridian 1 proprietary telephones.

LD 15 – Define Call Forward No Answer for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Attendant Console Options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ICI	xx CFN	Attendant Incoming Call Indicator for Call Forward No Answer, where: xx = key number (00-19).
TYPE	RDR	
- FNAD	(HNT) ATT FDN NO	Forward No Answer DID calls to the Hunt number. Forward No Answer DID calls to the attendant. Forward No Answer DID calls to the Flexible CFNA DN. No Answer DID calls are not forwarded.
- FNAT	(HNT) ATT FDN NO	Forward No Answer external calls to the Hunt number. Forward No Answer external calls to the attendant. Forward No Answer external calls to the Flexible CFNA DN. No answer external calls are not forwarded.
- FNAL	(HNT) ATT FDN NO	Forward No Answer local calls to the Hunt number. Forward No Answer local calls to the attendant. Forward No Answer local calls to the Flexible CFNA DN. No Answer local calls are not forwarded.
- CFNA	1-(4)-15	Number of ringing cycles before No Answer calls are forwarded (default is 4).
- CFN0	1-(4)-15	Number of normal rings for CFNA, Option 0. ***Note:*** CFNA has three ringing cycle options. Refer to the User Selectable Call Redirection feature in this NTP for more information.
- CFN1	1-(4)-15	Number of normal rings for CFNA, Option 1.
- CFN2	1-(4)-15	Number of normal rings for CFNA, Option 2.

LD 10 – Add/change Flexible Call Forward No Answer for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(FND) FNA	(Deny) allow Call Forward No Answer.
FTR	FDN xxxx…x	Flexible Call Forward No Answer DN (if the DN Expansion package is equipped, the DN can have up to 13 digits).

LD 11 – Add/change Flexible Call Forward No Answer for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FDN	xxx...x	Flexible Call Forward No Answer DN (if the DN expansion package is equipped, the DN can have up to seven digits).
CLS	(FND) FNA	(Deny) allow Call Forward No Answer.

LD 10 – Implement Call Forward No Answer to the Hunt DN on analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
HUNT	xxxx	Hunt DN where a No Answer call is to be routed (if the DN Expansion package is equipped, the DN can have up to 10 digits).
CLS	(FND) FNA	(Deny) allow CFNA.

LD 11 – Implement Call Forward No Answer to the Hunt DN on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(FND) FNA	(Deny) allow CFNA.
HUNT	xxxx	Hunt DN where a No Answer call is to be routed (if the DN Expansion package is equipped, the DN can have up to 10 digits).

Feature operation

No specific operating procedures are required to use this feature.

Call Forward Save on SYSLOAD

Contents

This section contains information on the following topics:

Feature description 691
Operating parameters 692
Feature interactions 692
Feature packaging 692
Feature implementation 692
Feature operation 692

Feature description

This feature enables sets to have their Call Forward (CFW) status saved as part of the data dump routine, thereby allowing the set to have its CFW status reinstated following a SYSLOAD. Whether a set has Call Forward activated following a SYSLOAD is dependant on the response to the Call Forward Save (CFWS) prompt in LD 17, and the status of the CFW as of the last successful data dump:

- If CFWS is set to NO (the default), no sets will have their CFW saved and all sets will have CFW set to the default (deactivated) following a SYSLOAD; or
- If CFWS is set to YES, all sets will have their CFW status saved and set to the state they were in as of the last successful data dump following a SYSLOAD.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Forward All Calls
Call Forward by Call Type

The Call Forward status of each telephone can be saved as part of the data dump routine and reinstated following a SYSLOAD operation.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 17 – Add or change Call Forward Save on data dump.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Systems Parameters
...		
MSCL	...	
- CFWS	(NO) YES	Call Forward Save on SYSLOAD.

Feature operation

If CFWS = YES, the Call Forward status of every set is saved at data dump. Should a SYSLOAD occur, all sets are returned to the Call Forward state that they were in as of the last successful data dump.

Call Forward to Trunk Restriction

Contents

This section contains information on the following topics:

Feature description 693
Operating parameters 693
Feature interactions 694
Feature packaging 694
Feature implementation 694
Feature operation 694

Feature description

The Call Forward to Trunk Restriction feature prevents stations from forwarding calls from their station to a Public Switched Telephone Network (PSTN) trunk. This conforms with the regulatory requirements of certain countries.

A second option of this feature allows calls to be forwarded to a PSTN trunk, as in previous operation, while recording the internal DN of the originating station (rather than the forwarding station) in the Call Detail Recording (CDR) record.

Operating parameters

The CDR option can only be applied to calls originated by internal telephones. Only the true originator's DN is recorded.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify the data blocks for trunk routes.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
...		
CFWR	(NO) YES	Call Forward Restriction (does not) does apply to the trunk route. If NO is the response, the IDOP prompt follows.
- IDOP	(NO) YES	Identify Originating Party. Responding YES modifies the trunk CDR for internal calls to identify the originating party instead of the forwarding station. If NO is the response, CDR is allowed to proceed as usual.

Feature operation

No specific operating procedures are required to use this feature.

Call Forward, Internal Calls

Contents

This section contains information on the following topics:

Feature description 695
Operating parameters 696
Feature interactions 697
Feature packaging 701
Feature implementation 701
Feature operation 703

Feature description

The Internal Call Forward (Internal CFW) feature allows you to selectively forward only internal calls to the Internal CFW DN.

Internal CFW is activated/deactivated on a per-telephone basis and is user programmable when Internal CFW is activated. On a Meridian 1 proprietary telephone, the Internal CFW feature (ICF) key is the only access method. On an analog (500/2500 type) telephone, Internal CFW can be accessed by either dialing SPRE and the Internal CFW feature code (9914), or by the appropriate Flexible Feature Codes (FFCs).

All internal calls terminating on the primary (or any single appearance) DN of an Internal CFW active telephone are automatically forwarded to the programmed Internal CFW DN (refer to the Operating parameters section for information on primary and secondary, and single and multiple appearance DNs).

An internal call is defined by the Internal CFW feature as one of the following:

- an extension-to-extension call
- a Direct Inward System Access (DISA) call
- an attendant-originated call
- a conference call
- a Group Call feature initiated call
- an incoming trunk call over a trunk route classified as internal (LD 16 where RCLS = INT), and
- an incoming Integrated Services Digital Network (ISDN) trunk call using private numbering.

Non-internal calls are not affected by the Internal CFW feature.

Operating parameters

Call Forward All Calls takes precedence over Internal CFW, but is not a prerequisite for the Internal CFW feature. For example, if a telephone is already CFW All Calls active, it will not be allowed to activate Internal CFW at the same time. Internal CFW can only be activated if CFW All Calls is explicitly deactivated.

Also, if Internal CFW is active when trying to activate CFW All Calls, Internal CFW will automatically be deactivated.

Internal CFW operation is consistent with the CFW All Calls feature. Therefore, when a Meridian 1 proprietary telephone activates Internal CFW, the following DNs will become Internal CFW activated:

- the primary DN (key 0), regardless of whether the DN is multiple appearance or not, and
- all secondary DNs that are single appearance.

Consequently, if all the appearances of a multiple appearance DN are on non-primary Meridian 1 proprietary telephone keys, calls to these DNs will never receive Internal CFW treatment.

When an analog (500/2500 type) telephone activates Internal CFW, regardless of whether the DN is multiple appearance or not, Internal CFW becomes activated.

Internal CFW supports only the voice line on digital telephones that have both voice and data options.

On 2317 and M3000 telephones, the CFW programming screen (invoked by pressing the CFW softkey), is not displayed when the ICF key is pressed. Instead, the screen displays the previously programmed ICF number.

If an Internal CFW call is rejected, a display message is given if the telephone is digital and has a digit display module (this display message is the same as that given to a failed CFW All Calls activation request). Otherwise, overflow tone is given.

Internal CFW is not maintained through a SYSLOAD.

Internal CFW is not supported on Basic Rate Interface (BRI) telephones.

Feature interactions

Attendant Administration

This feature does not support Internal CFW.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override Internal CFW. If the dialed DN of the set is idle, the DN can be blocked; if the DN is busy, busy tone will be heard.

Attendant Busy Verify

When the attendant is using this feature to call a telephone that is Internal CFW active, the call will not receive Internal CFW treatment.

Attendant-Extended Calls

When the attendant extends a call on its SRC key to a telephone that is Internal CFW active, the call on the SRC key will only receive Internal CFW treatment if it is an internal call.

Attendant Night Service

When a call to the attendant is redirected to the Attendant Night DN that is defined on an Internal CFW active telephone, the call will only receive Internal CFW treatment if it is an internal call.

Attendant Overflow

If Attendant Overflow redirects an internal call to a telephone that is Internal CFW active, the call will remain in the attendant queue, and will not receive Internal CFW treatment.

Call Forward All Calls
Call Forward Reminder Tone

If Call Forward Reminder Tone Allowed (CFRA) is in effect, whenever an analog (500/2500 type) telephone with Internal CFW active goes off hook to originate a call, the telephone sounds the reminder tone. The reminder tone is either Call Forward Dial Tone (CFDT) or Call Forward/Message Waiting Dial Tone (CFMW).

If the customer option is set to Call Forward Reminder Tone Denied (CFRD), whenever an analog (500/2500 type) telephone with internal CFW active goes off hook to originate a call, the telephone sounds either the normal dial tone (DIAL) or the Message Waiting Dial Tone (MWDT).

Call Forward, Break-In and Hunt Internal/External Network Wide

If a treated call is a transfer call and the transferring call is on the treating node, the transferred party will be considered. However, when the transferring party is not on the treating node, the transferring party will determine the treatment given.

Call Forward/Hunt Override Via Flexible Feature Code

Call Forward, Internal Calls is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Call Redirection by Time of Day

Call Forward Internal Calls takes precedence over Call Redirection by Time of Day.

Call Waiting
Call Waiting Redirection

Internal CFW takes precedence over Call Waiting and Call Waiting Redirection.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

If an incoming ISDN trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the outgoing trunk route on the tandem node also has CCP provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (i.e, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

Camp-On

Internal CFW takes precedence over Camp-On.

China – Attendant Monitor

If an attendant attempts to monitor a DN which is Call Forwarded and is idle, idle DN treatment is given.

China – Toll Call Loss Plan

Toll pad switching is also provided after call forwarding has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Do Not Disturb
Hunting

Internal Call Forward take precedence over Do Not Disturb and Hunting.

Flexible Voice/Data Terminal Number

Voice calls directed to a dynamic voice/data Terminal Number are forwarded, if either of these features are enabled. Data calls, to a dynamic voice/data TN, are not forwarded.

Network Intercom

Hot Type I calls respect or override all kinds of Call Forward features (Busy, No Answer, All Calls, Internal, etc.) according to per-set definitions. If Call Forward is respected, the call becomes a normally dialed call and the originator will receive the appropriate indication on their display.

Phantom Terminal Numbers (TNs)

Internal CFW cannot be enabled on a phantom TN.

Recorded Announcement for Calls Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) feature supports call forward to an outgoing external Central Office (CO) trunk if the trunk has the RANX flag set and is located in a node with a RAN trunk.

Recovery on Misoperation of Attendant Console

Call Forward takes precedence over the Misoperation feature.

Remote Call Forward

Remote CFW Activate (RCFA), Remote CFW Deactivate (RCFD), and Remote CFW Verify (RCFV) FFCs can be used only to access CFW All Calls; they cannot be used to access Internal CFW.

Trunk Barring

If an Originating Trunk Connection is forwarded to a barred route, it receives the intercept treatment specified in the Customer Data Block.

Feature packaging

Internal CFW requires the following packages:

- Basic Call Processing (BASIC) package 1 (CFW package required but does not have to be enabled)
- 500 Set Dial Access to Features (SS5) package 73 for access to analog (500/2500 type) telephones
- Flexible Feature Codes (FFC) package 139 to implement FFC

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Add/change Internal CFW for analog (500/2500 type) telephones.

2 LD 11 – Add/change CFW Internal Calls for Meridian 1 proprietary telephones.

3 LD 57 – Add/change Internal CFW for analog (500/2500 type) telephones using a Flexible Feature Code.

LD 10 – Add/change Internal CFW for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Telephone type.
FTR	ICF 4-(16)-23 XICF	Allow Internal CFW for the specified telephone and the maximum forward DN length. Remove Internal CFW from the telephone.

LD 11 – Add/change CFW Internal Calls for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
KEY	xx ICF 4-(16)-23 <nnnn>	Define an Internal CFW feature key for the telephone. The command consists of: xx = key number. ICF = feature mnemonic. 4-23 = the maximum forward DN length. nnnn = forward DN.
	xx null	Remove function/feature from a key.

LD 57 – Add/change Internal CFW for analog (500/2500 type) telephones using a Flexible Feature Code.

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove data.
TYPE	FFC	Flexible Feature Code.
CODE	ICFA ICFD ICFV	Access Code for Internal CFW Activate. Access Code for Internal CFW Deactivate. Access Code for Internal CFW Verify.
ICFA	xxxx	Internal CFW Activate code.
ICFD	xxxx	Internal CFW Deactivate code.
ICFV	xxxx	Internal CFW Verify code.

Feature operation

Meridian 1 proprietary telephone

To forward internal calls from a Meridian 1 proprietary telephone:

1 Press the ICF key.

2 Dial the number where calls are to be forwarded.

3 Press the ICF key.

To cancel Internal CFW from a Meridian 1 proprietary telephone:

- Press the ICF key.

Analog (500/2500 type) telephone

To forward internal calls from an analog (500/2500 type) telephone:

1 Lift the handset and dial SPRE 9914 (Internal CFW feature code)

– or –

Lift the handset and dial the Internal CFW Activate (ICFA) FFC.

2 Dial the number where calls are to be forwarded.

To cancel Internal CFW from an analog (500/2500 type) telephone:

- Lift the handset and dial SPRE 9914 (Internal CFW feature code)

– or –

Lift the handset and dial the Internal CFW Deactivate (ICFD) FFC.

Call Forward, Remote (Attendant and Network Wide)

Contents

This section contains information on the following topics:

Feature description 705
Operating parameters 706
Feature interactions 707
Feature packaging 709
Feature implementation 710
Feature operation 714

Feature description

Call Forward Remote (Attendant and Network wide) introduces the RCFW feature across the Meridian Customer Defined Network (MCDN), while also providing the attendant with RCFW capabilities. New ISDN FACILITY messages are used to facilitate the RCFW feature operation in an MCDN.

The feature capabilities of the set-based (Flexible Feature Code activated) network wide application of the RCFW feature match those of the current standalone RCFW feature.

The attendant RCFW functionality is controlled by a new flexible Attendant key (RFW). The attendant has the capability to view the current call forward number and determine the call forward status of any station. It is also possible for an attendant to activate or deactivate call forward for a particular station. This functionality is applicable both local within the system and network wide.

A new optional customer-based password is introduced for attendant RCFW operation. This password is configured in LD 15 and is the only password required for attendant RCFW operation. The station control password previously used by the Flexible Feature Code (FFC) set-based RCFW is not required when the attendant activates RCFW by pressing the RFW key.

Operating parameters

The network wide application of this feature is only applicable to nodes in an MCDN environment. The nodes in the network must be system switches. No other Central Office (CO) or circuit switched network type is supported for this feature operation.

For set-based network operation of the Remote Call Forward feature, the Station Control Password Length (SCPL) must be configured to be the same length for all nodes in the network. Attempts to operate RCFW with different SCPLs will result in overflow tone being presented to the user.

For network operation of the RCFW feature, the Private Network Identifier (PNI) must be configured consistently for all nodes in the network.

The Attendant and Network Wide RCFW features use the existing RCFW code to activate or deactivate call forward on stations. As such, all limitations applicable to the local RCFW feature are applicable to the network and attendant operation of the feature.

As the Swedish CD Attendant Console does not support alpha characters, the “PWD” prompt is not displayed on the console’s digit display when a password is required. The indication that a password is required is limited to the winking RFW key lamp.

No new hardware is required for this feature.

Feature interactions

Basic Rate Interface (BRI)

Since ISDN BRI sets do not support Flexible Feature Codes, Remote Call Forward cannot be activated from a BRI set.

Call Forward Activation from any Feature
Call Forward and Busy Status

There are no direct conflicts with either of these features and the RCFW feature.

Call Forward Destination Deactivation

Remote Call Forward (RCFW) and Call Forward Destination Deactivation (CFDD) provide the same functionality but are activated differently. CFDD does not require the call forward station's control password to deactivate the call forward functionality on the call forward station.

The call forwarded destination can use the Remote Call Forward deactivation FFC as well as CFDD to deactivate the Call Forward All Calls functionality on the call forward station.

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (that is, at the tandem node) from being displayed on the terminating set. In this case, the forwarding station must include the CPP in the forwarding DN (such, as *67 + ACOD + the DN on the terminating node).

Multiple Appearance Directory Numbers

The RCFW feature only applies to the primary appearances of Multiple Appearance DNs, and it is recommended that only one appearance of a Multiple Appearance DN be configured as the prime DN.

For the case of multiple stations with the same prime DN and SCPW, the RCFW operation will apply to the station that has the Multiple Appearance Redirection Prime (MARP) assigned to it.

If none of the stations having the DN and SCPW assigned are configured as the MARP TN for that DN, the RCFA and RCFD will apply to all stations matching the DN and SCPW.

The attendant-based RCFW feature will only apply remote call forward operation to the prime DN with MARP status. If the DN is not the prime DN or does not have MARP status, overflow tone will be received by the user.

Outpulsing of Asterisk and Octothorpe (OPAO)

If the OPAO package is equipped, the "#" will be treated as any other dialed digit and will not be used to signal end of dialing. The end of dialing digits to be used are defined in LD 15.

Phantom Terminal Numbers (TNs)

A Phantom TN does not physically exist, but can be configured with limited hardware associated with it (that is, no sets or line cards); however, all required data blocks are configured.

The Phantom TN feature uses the RCFW feature to configure and activate/deactivate the CFW DN on the Phantom TNs.

As the data blocks associated with Phantom TNs match those of standard analog (500/2500 type) telephones configured within the system, the operation of the RCFA and RCFD features on Phantom TNs is applicable to the RCFW feature. As such, the set-based local and network RCFW features can be used to configure and activate/deactivate the CFW DN of Phantom TNs.

The Phantom TN feature uses a Default Call Forward (DCFW) DN. If call forward is not active on the Phantom TN, all calls to the Phantom TN DN are routed to the DCFW DN.

The Phantom TN feature modifies the set-based RCFW feature so that if CFW is not active on the Phantom TN, and the CFW DN entered in the RCFV operation matches the DCFW DN, confirmation tone is returned to the RCFV user; if the CFW DN entered does not match the CDFW DN, overflow is returned.

This change to the set-based RCFV operation is applicable to the network RCFV operation. The operation of this feature network wide requires no changes to the ISDN message passing for the set-based network RCFV operation.

There is no Attendant RCFW operation which interacts with the DCFW DN of Phantom TNs.

Preventing Reciprocal Call Forward

When Preventing Reciprocal Call Forward Allowed (PVCA) is defined in LD 15, a set within the same customer configuration cannot be call forwarded to a set that is call forwarded back to it. Thus, CFW loops are prevented.

This feature applies when the CFW DN is changed by Remote Call Forward. For network operation of the set- and attendant-based RCFW features, entering an invalid CFW DN (under the rules of the PRCF feature) results in overflow tone being returned and the CFW DN being ignored.

Traffic Measurements

The peg count, for the attendant RFW key, will be generated on the first RFW key press of the RCFW operation. While the RFW key may be pressed multiple times during a single RCFW function, the peg count will only be implemented once.

The RFW key peg count will be included in the TFC005 feature key usage traffic report.

Feature packaging

The Attendant Remote Call Forward (ARFW) package 253 must be provisioned to activate the Attendant-based RCFW feature.

For network operation the following software packages are required:

- Integrated Services Digital Network (ISDN) package 145
- Network Alternate Route Selection (NARS) package 58
- Any other trunk or dialing plan packages, as required by the customer's configuration

The following are prerequisites for set-based RCFW:

- Optional Features (OPFT) package 1
- Flexible Feature Codes (FCC) package 139
- Controlled Class of Service (CCOS) package 81

The following are prerequisites for implementation on analog (500/2500 type) telephones:

- Special Service for 2500 Sets (SS25) package 18
- 500 Set Dial Access to Features (SS5) package 73

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Set the Station Control Password Length.

2 LD 15 – Configure a Special Prefix Number (SPRE) for the customer.

3 LD 57 – Define Remote Call Forward FFCs and set FFCT.

4 LD 10 – Set the Station Control Password and allow Call Forward.

5 LD 11 – Set the Station Control Password and allow Call Forward.

6 LD 12 – Configure the Attendant Console RFW key.

7 LD 15 – Configure the Attendant RCFW password.

Set-based configuration

LD 15 – Set the Station Control Password Length.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FFC	Flexible Feature Code

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- SCPL	0-8	Station Control Password Length (must be consistent network wide).
- FFCS	YES	Change end of dialing digits in FFC.
- - STRL	1-3	Number of digits to indicate FFC end of feature activation.
- - STRG	(#) xxx	One to three digits to indicate FFC end of a feature activation.

LD 15 – Configure a Special Prefix Number (SPRE) for the customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- SPRE	xxx	Special Prefix Number.

LD 57 – Define Remote Call Forward FFCs and set FFCT.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Confirmation tone is (is not) to be given after an FFC.
CODE	RCFA	Remote Call Forward Activate.
RCFA	xx	xx = RCFA code.

CODE	RCFD	Remote Call Forward Deactivate.
RCFD	xx	xx = RCFD code.
CODE	RCFV	Remote Call Forward Verify.
RCFV	xx	xx = RCFV code.

LD 10 – Set the Station Control Password and allow Call Forward.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
FTR	CFW 4-(16)-23	Allow Call Forwarding and set the forwarding DN length.

LD 11 – Set the Station Control Password and allow Call Forward.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCPW	xxxxxxxx	Station Control Password (0 to 8 digits, defined in LD 15).
KEY	xx CFW 4-(16)-23	Assign Call Forward key (xx) and set the forwarding DN length.

Attendant-based configuration

A new Flexible Attendant feature key, RFW, has been added to this overlay. Configuration of the key on the Attendant Console is required to allow attendant access to the RCFW feature. Configuration of the RFW key is only allowed if the ARFW package is equipped.

LD 12 – Configure the Attendant Console RFW key.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	ATT 1250 2250	Attendant type – the RFW key can be configured on QCW4, M1250, M2250 and Swedish CD Attendant Consoles.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
KEY	xx RFW	Key number assigned as Attendant Remote Call Forward key.

LD 15 – Configure the Attendant RCFW password.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT	Attendant console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- IRFR	(NO) YES	Internal Remote Call Forward Password required. ***Note:*** An internal password is the password required to perform an attendant RCFW operation within the same customer as the attendant.

- - IRFP	xxxxxxxx	Internal RCFW Password (only prompted if the response to IRFR is YES). The password length is one to eight digits; the password is numeric only.
- XRFR	(NO) YES	External Remote Call Forward Password required. ***Note:*** An external password is the password required to perform an attendant RCFW operation on a different customer than the attendant.
- XRFP	xxxxxxxx	External RCFW password (only prompted if the response to XRFR is YES). The password length is one to eight digits; the password is numeric only.

Feature operation

Network Wide Set-based Remote Call Forward

From the remote set dial:

1 FCC RCFA code.

2 SCPW for the set to be forwarded.

3 The complete DN of the set to be forwarded. This DN is the full DN required to call the set to be forwarded from the user's present location.

Expected Result: Confirmation tone is delivered to the user.

From the remote set continue dialing:

4 The CFW DN to be activated followed by the end of dial indicator (#).

Expected Result: Confirmation tone is delivered to the user.

Error Condition: If an error should occur during set-based RCFW, the user will be presented with an Overflow tone. To remove the error condition, the user must release from the operation and begin again.

Attendant-based Remote Call Forward

From the Attendant Console, perform the following:

1 Press an idle loop key followed by the RFW key.

Expected Result: The RFW key is flashing and the Loop key is steady lit.

2 Dial the DN of the set to be forwarded.

Expected Result: If a password is required, the RFW key is winking, and the console display shows “PWD –”. If the console does not support alpha characters, the display will be blank.

If a password is not required, the console display will show the DN of the set to be forwarded followed by the CFW DN stored on that set. The RFW key lamp will display the status of the CFW DN. If the RFW lamp is flashing, CFW is not active; if the RFW lamp is steady lit, CFW is active. Proceed to Step 4.

3 Dial the password followed by #.

Expected Result: The console display will show the DN of the set to be forwarded followed by the CFW DN stored on that set. The RFW key lamp will display the status of the CFW DN. If the RFW lamp is flashing, CFW is not active; if the RFW lamp is steady lit, CFW is active.

4 The user can now enter a new CFW DN or press the RFW key to activate or deactivate the stored CFW DN.

Expected Result: The console display will show the DN of the set to be forwarded followed by the CFW DN. If the RFW lamp is flashing, CFW is not active; if the RFW lamp is steady lit, CFW is active.

5 When RCFW operation is in this state, the user has the following three options:

- Press the Release or Release Source key to terminate RCFW operation.
- Press the RFW key to reverse the CFW status.
- Enter a new CFW DN to begin the task of changing the CFW DN programmed. The new CFW DN is not active until the RFW key is pressed again.

Error Condition: If an error should occur during the attendant-based RCFW, the user will be presented with an Overflow tone. To remove the error condition, the user must release from the operation and begin again.

Call Forward/Hunt Override Via Flexible Feature Code

Contents

The following are topics in this section:

Feature description 717
Operating parameters 718
Feature interactions 718
Feature packaging 723
Feature implementation 724
Feature operation 726

Feature description

Call Forward Override provides all telephone users (having a specific Class of Service) and attendants with the ability to override Intercept Computer Call Forward (ICP-CFW), Call Forward All Calls, Call Forward No Answer, Hunting and Make Set Busy by entering a Flexible Feature Code. In order to use this feature, the originating set must have Call Forward Hunt Allowed (CFHA) Class of Service. When this feature is enabled if override is attempted, and the called party is idle, the set is rung regardless of any diversion. If the dialed set is busy and has Hunt active, the calling party will terminate on the wanted set and receive a busy indication. Sets without Call Forward/Hunt Override denied (CFHD) Class of Service will not be able to use the Call Forward/Hunt Override (CFHO) Via Flexible Feature Code (FFC) feature.

Call Forward/Hunt Override Via FFC works in network environments with system nodes and Meridian Customer Defined Network (MCDN) links.

Operating parameters

The Call Forward/Hunt Override FFC can only be used in predial mode from a set (for instance, it must be dialed before dialing the DN that has Call Forward All Calls, Intercept Call Forward, Call Forward No Answer, Internal Call Forward, Hunt, or Make Set Busy active).

The Call Forward/Hunt Override FFC can only be dialed from its own node (that is, it must be dialed before any trunk access code).

On an ABCD set the Call Forward/Hunt Override FFC can only be configured as a predial FFC (ABCD sets are a type of German telephone).

Call Forward/Hunt Override FFC can only be used against extensions with one of the following type: HOT/MCN/MCR/SCN/SCR/Basic Rate Interface (BRI) DNs and analog (500/2500 type) telephones.

It is not possible for BRI extensions to dial a Call/Forward Hunt Override FFC.

The Call Forward/Hunt Override Via FFC feature can only be used in standalone and MCDN environments. If no MCDN links are involved, no information about Call Forward/Hunt Override will be passed on to other nodes.

To get the functionality of Call Forward/Hunt Override Via FFC in an MCDN environment these enhancements must be integrated in the originating node, terminating node and any intermediate nodes.

Feature interactions

Attendant Blocking of DN

Using Call Forward/Hunt Override FFC after activation of ABDN is not allowed. Any attempt will be canceled and overflow tone will be returned.

Automatic Call Distribution

Automatic Call Distribution (ACD) DNs are not overridden by Call Forward/Hunt Override Via FFC. Any attempt will be canceled and access denied treatment will be returned. Individual DNs on an ACD set are overridden by Call Forward/Hunt Override Via FFC with the same limitations as for other sets.

Attendant Barge-in
Attendant Busy Verify
Attendant Break-in

Using Call Forward/Hunt Override Via FFC after activation of Barge-in, Busy Verify or Break-in is not allowed. Attempts will be canceled and overflow tone will be returned.

Using post-dial Break-in after dialing the Call Forward/Hunt Override FFC is possible after encountering a busy set, if Break-in is enabled.

Basic Rate Interface (BRI)

BRI sets are not supported; any attempt to dial Call Forward/Hunt Override from a BRI set will be ignored and access denied treatment will be returned.

BRIT

BRI TIE trunks in a Meridian Customer Defined Network (MCDN) are supported.

Call Forward All Calls
Call Forward No Answer
Call Forward and Busy Status
Call Forward, Internal Calls
Call Forward No Answer/Flexible Call Forward No Answer
Make Set Busy
Secretarial Filtering

These features are overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the features themselves.

Call Redirection by Time of Day

Call Forward/Hunt Override Via FFC has precedence over Call Redirection by Time of Day.

Call Transfer

A set can activate Call Forward/Hunt Override Via FFC when initiating a transfer. If the transfer is completed while ringing, the Call Forward/Hunt Override will still be active and passed on to the transferred party.

Call Waiting

Call Waiting can be used even if the Call Forward/Hunt Override Via FFC feature has been activated. When a busy set with Call Waiting configured is encountered, it will terminate on Call Waiting.

Call Waiting Redirection

There is no interaction with the Call Waiting treatment component of the Call Waiting Redirection feature. However, Call Forward/Hunt Override Via Flexible Feature Code does override CFNA, and thus the CFNA treatment given to unanswered Call Waiting calls by the Call Waiting Redirection feature is overridden by the CFHO feature. The incoming call will continue to be given Call Waiting treatment as if the Call Waiting Redirection feature is disabled when the CFHO feature is enabled by the calling party.

Camp-on

When a busy set is encountered, it is possible to Camp-on to the set, even if Call Forward/Hunt Override Via FFC has been activated.

Digital Private Network Signaling System One (DPNSS1)

DPNSS1 is only supported as an incoming trunk transferred to a MCDN environment using Call Forward/Hunt Override Via FFC.

Direct Inward System Access

Direct Inward System Access is not supported. Any attempt to dial the Call Forward/Hunt Override FFC will be ignored and access denied treatment will be returned.

Do Not Disturb

Do Not Disturb is not overridden by the Call Forward/Hunt Override Via FFC feature.

Flexible DN
External Flexible DN

It is not possible to store the Call Forward/Hunt Override FFC as a Flexible Directory Number or External Flexible Directory Number.

Group Call

It is not possible to use the Call Forward/Hunt Override FFC as a Group Call DN.

Group Hunt

Primary Line Directory Numbers (PLDNs) are not overridden by the Call Forward/Hunt Override Via FFC feature. Any attempt will be ignored and access denied treatment will result.

Hunt

This feature is overridden by the Call Forward/Hunt Override Via FFC feature. If a set activating Call Forward/Hunt Override Via FFC encounters a busy set no hunt steps will be performed; the call will terminate on the DN and busy tone will be returned.

Hunt DN/External Hunt DN

It is not possible to store the Call Forward/Hunt Override FFC as a Hunt or External Hunt DN.

Idle Extension Notification

This feature can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When a busy set is encountered, it is possible to place an Idle Extension Notification request against the set.

Intercept Computer Call Forward

This feature is overridden by the Call Forward/Hunt Override Via FFC feature. The Call Forward/Hunt Override FFC replaces the Intercept Computer Override FFC.

Intercept Computer Dial from Directory - Pre-dial Operations

Call Forward Hunt Override Via Flexible Feature Code can be dialed prior to dialing the DN from the Intercept Computer.

Last Number Redial

The Call Forward/Hunt Override FFC and the dialed DN are stored under Last Number Redial.

Multiple Appearance Multiple Call Arrangements (MCAs) Multiple Appearance Single Call Arrangements (SCAs)

If the Call Forward/Hunt Override FFC is used against an MCA (MCR/MCN) or SCA (SCR/SCN) DN it will override any active forward and terminate on all idle appearances. If all appearances are busy, busy treatment will be returned.

Primary Line Directory Number (PLDN)

It is not possible to store the Call Forward/Hunt Override FFC as a PLDN.

Phantom DN

Phantom TN

These features are not overridden by the Call Forward/Hunt Override Via FFC feature. If Call Forward/Hunt Override Via FFC is used against a phantom TN the call will be canceled and overflow tone will be given.

Priority Override

It is possible to use Priority Override after using the Call Forward/Hunt Override FFC and encountering a busy set.

Radio Paging

If Radio Paging is activated in a call where Call Forward/Hunt Override has been used, the Call Forward/Hunt Override feature will be deactivated.

Ring Again
Network Ring Again

Using the Ring Again feature is possible after using the Call Forward/Hunt Override FFC and encountering a busy signal. Ring Again can be placed against the set for which the Call Forward/Hunt Override FFC was used (that is, the set with CFW active should be rung by the Ring Again feature).

Ring Again No Answer
Network Ring Again No Answer

Using the Ring Again No Answer feature is possible after using the Call Forward/Hunt Override FFC and encountering an idle set that does not answer. Ring Again No Answer can be placed against the set for which the Call Forward/Hunt Override FFC was used (that is, the set should be rung by the Ring Again No Answer feature).

Single Digit Access

It is not possible to store Call Forward/Hunt Override FFCs in a Single Digit Access list.

Semi-automatic Camp-On

This feature can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When encountering a busy set, it is possible to activate Semi-automatic Camp-On, if it is applicable.

Speed Call

The Call Forward/Hunt Override FFC cannot be stored in a speed call list.

Feature packaging

In a standalone environment, the Flexible Feature Codes (FFC) software package 139 must be provisioned to activate this feature.

For network environments, Network Attendant Service (NAS) package 159 must also be provisioned. Attendant Overflow Position (AOP) package 56 must be restricted, as it is mutually exclusive with Network Attendant Service.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 57 – Define FFC for Call Forward/Hunt Override.

2 LD 10 – Set Class of Service for the Forward/Hunt Override Via FFC feature for analog (500/2500 type) telephones.

3 LD 11 – Set Class of Service for the Forward/Hunt Override Via FFC feature for Meridian 1 proprietary telephones.

4 LD 18 – Configure ABCD key for the Forward/Hunt Override Via FFC feature.

LD 57 – Define FFC for Call Forward/Hunt Override.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FFC	Flexible Feature Code.
...		
CODE	CFHO	Call Forward/Hunt Override Via FFC.
CFHO	nnnn	Call Forward/Hunt Override FFC.

LD 10 – Set Class of Service for the Forward/Hunt Override Via FFC feature for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Type of telephone set.
...		
CLS	(CFHD) CFHA	Call Forward/Hunt Override Via FFC is (denied) or allowed.

LD 11 – Set Class of Service for the Forward/Hunt Override Via FFC feature for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Type of telephone set.
...		
CLS	(CFHD) CFHA	Call Forward/Hunt Override Via FFC is (denied) or allowed.

LD 18 – Configure ABCD key for the Forward/Hunt Override Via FFC feature.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	ABCD	Modifying 16-button DTMF.
...		
PRED	YES	Function table for pre-dial.
A	CFHO	CFHO is assigned to key A.

B	CFHO	CFHO is assigned to key B.
C	CFHO	CFHO is assigned to key C.
D	CFHO	CFHO is assigned to key D.

Feature operation

Standalone

To activate the Call Forward/Hunt Override feature, the user dials the FFC for Call Forward/Hunt Override and the DN of the wanted party. If the set is idle, the set is rung regardless of any diversion (for example, Call Forward All Calls, Intercept Call Forward, Call Forward No Answer, or Hunt) or Make Set Busy on the set.

If the set(s) have displays, the display(s) are updated. If the display on the originating set is updated when the call is answered, the Call Forward/Hunt Override FFC will no longer be displayed.

If the dialed set is busy and Hunt is active, the calling party will terminate on the wanted set and will receive busy indication.

If the dialed set is idle, but does not answer within the defined number of ringing cycles for CFNA, the call is not forwarded (that is, it continues to ring).

If the dialed set is busy, the attendant can activate Camp-on, if Camp-on is applicable. In addition, Ring Again can be placed against a set for which Call Forward/Hunt Override was used and a busy set was encountered.

Network

The user gets the same functionality in a Meridian Customer Defined Network (MCDN) as in standalone environments. The Call Forward/Hunt Override information is transmitted from the originating node to the terminating node using the Network Attendant Service (NAS) feature.

Activation of the service is call dependent; network-wide Call Forward/Hunt Override is part of the NAS feature.

Call Hold, Deluxe

Contents

This section contains information on the following topics:

Feature description 727
Operating parameters 728
Feature interactions 728
Feature packaging 730
Feature implementation 730
Feature operation 731

Feature description

Deluxe Call Hold (DHLD) offers two options: Individual Hold and Exclusive Hold.

Individual Hold indicates only those calls placed on hold on Meridian 1 proprietary telephones in a multiple appearance, single call arrangement. When a user puts a call on hold, normal hold (winking) is indicated at the user's telephone only. A slow flicker is shown at all other telephones with the multiple appearance.

With Exclusive Hold Class of Service, multiple appearances of a line remain exclusive to the user's telephone, even when the call is put on hold. While hold (winking) is indicated at the telephone holding the call, the Directory Number (DN) lamp is steadily lit on all other appearances of the held call. The Privacy Release key must be used for access by other appearances of the DN. Telephones with the Exclusive Hold capability can be held at any single-line or Meridian digital telephone with an appearance.

Operating parameters

Exclusive Hold has priority over Individual Hold. If a telephone is equipped with Exclusive Hold, the other telephones receive the Exclusive, not Individual, Hold indication.

Feature interactions

Attendant Administration

Deluxe Hold (DHLD) cannot be administered through the Attendant Administration feature.

Attendant Break-In

The attendant cannot break in to a call on hold.

Camp-On, Forced
Override, Enhanced
Override, Priority

Neither held calls nor telephones with calls on hold can be camped on or overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-on or Priority Override.

Call Hold, Deluxe

When a call is retrieved from hold, the calling and called parties' displays reflect their individual DPD Class of Service options.

Call Park on Unsupervised Trunks

A Disconnect Timer applies to held calls on all trunks on the route. All answered calls in the held state will be disconnected if left in that state for an extended period.

Call Party Name Display

When a call is put on hold, the holding telephone's display clears. The held telephone's display does not change. When the telephone reestablishes the call, the display returns the original DN and name.

Call Transfer

A consultation call can be placed on Hold.

Called Party Control on Internal Calls

The calling party and called party can put either party on hold. However, the calling party cannot release the call while the called party is on hold. The called party is permitted to release the call.

Calling Party Privacy

When a user takes an incoming trunk call with the Privacy Indicator off of hold, no Calling Party Number or Name will be displayed on the set.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is denied if the requested party is put on hold by another station at the same node. This restriction also applies to the unrequested party if the unrequested party is located at the same node as the requested party (standalone) or if the requested party and the unrequested party are linked using DPNSS1.

Mixed DNs

If a call is put on Exclusive Hold in a mixed Directory Number (DN) group, other telephones with an appearance of the DN that go off hook are not included in the call, nor do they receive any tone. Privacy Release cannot be used with exclusively held calls in a mixed-appearance DN group.

Multiple Appearance Directory Number

If two or more proprietary sets of the same Directory Number are in conference due to privacy release or privacy override, then only one set can hold the call at a given time.

Music, Enhanced

A caller placed on Hold by a member of a multiple appearance group receives Music regardless of whether the call is on Hold or Exclusive Hold.

Predictive Dialing

If an established call is put on hold by the set initiating the Fast Transfer, the switch will not be able to transfer the call. The switch can only transfer a call if it is in the established state.

Feature packaging

Deluxe Hold (DHLD) package 71 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable/disable Individual Hold for the customer.

2 LD 10 – Enable/disable Exclusive Hold for analog (500/2500 type) telephones.

3 LD 11 – Enable/disable Exclusive Hold for Meridian 1 proprietary telephones.

LD 15 – Enable/disable Individual Hold for the customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.

TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(IHD) IHA	(Disable) enable Individual Hold.

LD 10 – Enable/disable Exclusive Hold for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XHD) XHA	(Disable) enable Exclusive Hold.

LD 11 – Enable/disable Exclusive Hold for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XHD) XHA	(Disable) enable Exclusive Hold.

Feature operation

No specific operating procedures are required to use this feature.

Call Hold, Individual Hold Enhancement

Contents

This section contains information on the following topics:

Feature description 733
Operating parameters 735
Feature interactions 735
Feature packaging 737
Feature implementation 738
Feature operation 740

Feature description

Individual Hold Enhancement (IHE) expands the functionality of the Individual Hold feature. This enhancement is part of Deluxe Call Hold. Individual Hold Enhancement provides, in a single line Multiple Appearance Directory Number (MADN) environment, the following options:

- Lamp Option
- Release Option

With the Lamp Option allowed (HLPA), if one of the single line MADNs is on hold, all other appearances of the same single line MADN now remain steadily lit.

With the Lamp Option denied (HLPD), the existing Individual Hold functionality is retained.

With the Release Option allowed (HRLA), if the user presses the Hold key while another member of the same single line MADN is still active on the call, the set that put the call on hold is now disconnected from the call.

With the Release Option allowed (HRLA), if the user is the only active member of the single line MADN on the call, pressing the hold key puts the call on hold as per the existing operation.

With the Release Option denied (HRLD), the existing Individual Hold functionality is retained.

Table 31 shows the lamp status of a single line MADN when the Lamp (HLPA/HLPD) and Release (HRLA/HRLD) Options are configured in LD 15.

Table 31
Lamp Status of a single line MADN with Lamp (HLPA/HLPD) and Release (HRLA/HRLD) Options configured

OPT in LD 15	Held or Disconnected Appearance	Other Appearances
IHA, HLPD, HRLD (See Note 1)	wink (held appearance)	flicker
IHA, HLPA, HRLD	wink (held appearance)	steadily lit
IHA, HLPD, HRLA (See Note 2)	steadily lit (disconnected appearance)	steadily lit
IHA, HLPD, HRLA (See Note 3)	wink (held appearance)	flicker
IHA, HLPA, HRLA (See Note 2)	steadily lit (disconnected appearance)	steadily lit
IHA, HLPA, HRLA (See Note 3)	wink (held appearance)	steadily lit

Note 1: In this situation, the existing functionality of Individual Hold is retained.

Note 2: In this situation, a single line MADN member is disconnected from the active call. Another member of the same single line MADN remains active on the call.

Note 3: In this situation, a single line MADN member places a call on hold. No other member of the same single line MADN is active on the call.

Operating parameters

For Individual Hold Enhancement to be activated, Individual Hold Allowed (IHA) and the Lamp (HLPA/HLPD) and Release (HRLA/HRLD) Options must be defined in the Customer Data Block. Also, a single line MADN must be defined for the sets.

The Lamp Option applies to Meridian 1 proprietary sets with a call on hold and to analog (500/2500 type) sets which perform the Permanent Hold operation.

If the single line MADN member is an analog (500/2500 type) set, the lamp option only applies when Permanent Hold is initiated from that set.

The Release Option only applies when two or more parties with the same single line MADN are active in a conference call.

The Release Option (HRLA/HRLD) only applies to Meridian 1 proprietary sets with a Hold key or to Meridian 1 proprietary sets with Automatic Hold enabled (CLS = AHA) in LD 11.

For Meridian 1 proprietary sets with the Release Option allowed (HRLA), the Conference (A03/A06) and Transfer (TRN) keys are ignored if more than one single line MADN is active on a Conference call.

For analog (500/2500 type) sets, Transfer, Conference, and Permanent Hold work as per the existing operation.

Feature interactions

Automatic Call Distribution

When an Individual Directory Number (IDN) on an Automatic Call Distribution (ACD) set is configured as a single line MADN, both the Lamp and Release Options of the Individual Hold Enhancement feature are applicable to that IDN.

Automatic Hold

When a proprietary set has Automatic Hold allowed and more than one single line MADN (SCR/SCN/HOT/PVR/PVN) is active on a conference call, if the user presses the hold key or presses the active single line MADN, the following occurs:

- Without the Release Option enabled, the active call on the single line MADN is put on hold. That is, the lamp on the single line MADN flashes as per the existing operation.
- With the Release Option allowed (HRLA), the active call on the single line MADN is disconnected.

If only one single line MADN is active on a conference call, the existing operation is retained.

Multi-Party Operations

Call Join

If a conference is set up using Call Join, Individual Hold Enhancement still functions.

When the Lamp Option is allowed (HLPA) and the user presses the Hold key on an active single line MADN in Call Join operation, the lamps of all other appearances of the same single line MADN are steadily lit.

When the Release Option is allowed (HRLA) and the user presses the Hold key on an active single line MADN in order to enlarge a conference using the Call Join feature, the following occurs:

- The call is disconnected if another member of the same single line MADN is still active on the call.
- The call is placed on hold if the active single line MADN is the only active member of the single line MADN on the call.

Conference (A03/A06, C6A)

With the Release Option allowed (HRLA), the Conference key is ignored on the Meridian 1 proprietary set, when more than one Multiple Appearance Directory Number (MADN) is active in the conference.

Exclusive Hold

The Exclusive Hold feature takes precedence over the Lamp Option of the Individual Hold Enhancement feature. The lamps on all other single line appearances of the MADN are steadily lit while the call is held on one of the single line MADNs. When the Lamp Option is enabled (HLPA), the existing Exclusive Hold functionality is retained.

With the Release Option allowed (HRLA) and more than one single line MADN active in a conference, the call is disconnected when the hold key is pressed.

Permanent Hold on an analog (500/2500 type) set

When an analog (500/2500 type) set puts a call on Permanent Hold, the lamp status remains steadily lit on all other members of the same single line MADN if the customer has the Lamp Option allowed.

With the Release Option configured, if an analog (500/2500 type) set with the same single line MADN initiates Permanent Hold, the call is not dropped even if other appearances are active on this call. This is as per the existing functionality.

Switch Hook Flash

When a single line MADN member uses Switch Hook Flash to place a call on hold, all other appearances of the same single line MADN remain lit as per the existing operation.

When an analog (500/2500 type) set with the same single line MADN initiates hold by Switch Hook Flash, the existing functionality is retained.

Transfer (TRN)

For Meridian 1 proprietary sets with the Release Option allowed (HRLA), the transfer feature is disabled on the single line MADN, while more than one appearance is active in the conference.

For analog (500/2500 type) sets, transfers work as per the existing operation.

Feature packaging

Individual Hold Enhancement requires Deluxe Call Hold (DHLD) package 71.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Configure Individual Hold Allowed, the Individual Hold Lamp Option and the Individual Hold Release Option in the Customer Data Block.
2. LD 10 – Enable Call Transfer Allowed (XFA), Enhanced Hot Line Denied (EHTD), Exclusive Hold Denied (XHD), and Permanent Hold (PHD) for analog telephones.
3. LD 11 – Enable Privacy Override Allowed (POA) and Exclusive Hold Denied (XHD) for Meridian proprietary telephones.

LD 15 – Configure Individual Hold Allowed, the Individual Hold Lamp Option and the Individual Hold Release Option in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options data.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
OPT	IHA	Individual Hold Allowed.
	HLPA	Individual Hold Lamp Option Allowed. HLPD = Individual Hold Lamp Option Denied (default).
	HRLA	Individual Hold Release Option Allowed. HRLD = Individual Hold Release Option Denied (default).
...		

LD 10 – Enable Call Transfer Allowed (XFA), Enhanced Hot Line Denied (EHTD), Exclusive Hold Denied (XHD), and Permanent Hold (PHD) for analog telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Analog (500/2500 type) set data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	XFA EHTD XHD	Call Transfer Allowed. Enhanced Hot Line Denied. Exclusive Hold Denied.
...		
FTR	PHD	Permanent Hold.
...		

LD 11 – Enable Privacy Override Allowed (POA) and Exclusive Hold Denied (XHD) for Meridian proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type where xxxx is: SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2616, 2317, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	POA XHD	Privacy Override Allowed. Exclusive Hold Denied.

...		
KEY	xx SCR yyyy	Single Call Ringing key, where: xx = key number SCR = Single Call Ringing yyyy = DN KEY may also be defined as HOT (Two-way), PVN, PVR, or SCN.

Feature operation

With the Release Option allowed (HRLA), the user of a single line MADN disconnects from an active call by

- pressing the release key; or
- pressing the hold key to disconnect from the active call while another member of the same single line MADN is still active on the call.

However, if a user is the only active member of the single line MADN on the call with the Release Option allowed (HRLA), pressing the hold key puts the call on hold as per the existing operation.

With the Release Option denied (HRLD), the existing functionality is retained. Therefore, the user must press the release key to disconnect from the active call. When the hold key is pressed, the call is not released.

Call Hold, Permanent

Contents

This section contains information on the following topics:

Feature description 741
Operating parameters 741
Feature interactions 742
Feature packaging 744
Feature implementation 744
Feature operation 745

Feature description

Permanent Hold holds an active call on a 2500 telephone without attendant assistance. Calls cannot be originated or received while in the Permanent Hold mode. Incoming calls receive a busy signal if Hunting is not defined for the called telephone.

If the telephone user goes on hook after activating Permanent Hold, the telephone periodically receives a one-second ring burst as a reminder that the call is on hold. This interval is defined at the customer level.

Operating parameters

Permanent Hold is allowed only when a call is active and if the Class of Service allows transfer.

If Busy Verify is attempted on a telephone with a call on Permanent Hold, busy tone is received.

Override cannot be used on a telephone with a call on Permanent Hold.

Permanent Hold cannot be activated during a Conference call.

Two parties, connected trunk to trunk, can activate Permanent Hold at the same time if they both have the feature defined. After being placed on Permanent Hold, the second party can flash the switchhook and dial #4 to hold the call. After flashing the switchhook, any dialing sequence other than the access code results in overflow tone.

Permanent Hold is not supported on station-to-station calls.

If the telephone activating Permanent Hold is part of a mixed arrangement with another 2500, or Meridian 1 proprietary telephone, the following events occur:

- If a different telephone with the same DN goes off hook, that telephone connects to the held party.
- When Permanent Hold is activated, the DN lamp on the Meridian 1 proprietary telephone remains steadily lit.

If the telephone activating Permanent Hold goes off hook, it is automatically reconnected to the held call.

If the held party disconnects, the hold reminder ring stops.

Feature interactions

Attendant Break-In

The attendant cannot break in to a call on hold.

Audible Reminder of Held Call (ARCH)

If Audible Reminder of Held Call (ARCH) is enabled in LD 15, the Audible Reminder of Held Call (ARCH) timer takes precedence over the Permanent Hold timer.

AC15 Recall: Timed Reminder Recall

Call Hold Permanent is activated when the attendant presses the HOLD key then the Release (RLS) key when extending a call, the call will then be permanently held on the Loop key. If the attendant retrieves the original call on hold by pressing the Loop key, the recall timer is stopped. If the attendant then presses the RLS key, the call is extended and the recall timer is restarted.

Call Park on Unsupervised Trunks

A Disconnect Timer applies to held calls on all trunks on the route. All answered calls in the held state will be disconnected if left in that state for an extended period.

Calling Party Privacy

When a user takes an incoming trunk call with the Privacy Indicator off of hold, no Calling Party Number or Name will be displayed on the set.

Camp-On, Forced Override, Enhanced Override, Priority

Neither held calls nor telephones with calls on hold may be camped on or overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-on or Priority Override.

China – Attendant Monitor

Monitoring is not affected if anybody involved in the monitor's call activates hold, except for the case of a simple call. For a monitored simple call, activating hold deactivates monitoring. In all cases, activation of music on hold deactivates monitoring.

An attendant monitoring a call cannot put the monitored DN on hold. The attendant pressing the hold key has no effect while monitoring is enabled.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is denied if the requested party is put on hold by another station at the same node. This restriction also applies to the unrequested party if the unrequested party is located at the same node as the requested party (standalone) or if the requested party and the unrequested party are linked using DPNSS1.

Predictive Dialing

If an established call is put on hold by the set initiating the Fast Transfer, the switch will not be able to transfer the call. The switch can only transfer a call if it is in the established state.

Privacy

A call placed on Permanent Hold has Privacy removed. Privacy is reinstated when the call is removed from Permanent Hold.

Feature packaging

Special Service for 2500 Sets (SS25) package 18 includes Permanent Hold and has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Enable/disable Permanent Hold reminder ring timer for the customer.
2. LD 10 – Enable/disable Permanent Hold for 2500 telephones.

LD 15 – Enable/disable Permanent Hold reminder ring timer for the customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.

TYPE:	TIM	Timers
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- PHDT	1-(30)-63	Permanent Hold reminder ring timing in two-second increments (that is, 30 = 60 seconds).

LD 10 – Enable/disable Permanent Hold for 2500 telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	XFA	Allow transfer.
FTR	PHD	Enable Permanent Hold.

Feature operation

To place a call on hold, follow these steps:

- While on an active call, flash the switchhook or press the Link key.
- Dial #4, or the Flexible Feature Code (FFC), if enabled.
- Hang up.
  The Permanent Hold timer begins.

To retrieve a held call, lift the handset.

Call Park

Contents

This section contains information on the following topics:

Feature description 747
Operating parameters 749
Feature interactions 750
Feature packaging 757
Feature implementation 758
Feature operation 761

Feature description

Call Park (CPRK) places a call in a parked state, similar to hold, where it can be retrieved by any Attendant Console or telephone. A parked call must have an access ID, also known as a Park DN. This is done by parking the call on a System Park DN or on any telephone Directory Number (DN) in the system. A parked call does not occupy a DN, nor is there a lamp to indicate its presence.

Up to 50 System Park DNs are available per customer. There is no limit to the number of DNs that can be used as a Call Park access ID. However, only one call at a time can be parked against any particular telephone or System Park DN.

In addition, the system can offer a default access ID. If System Call Park is defined, the default access ID for the following equipment is the next available System Park DN for the equipment:

- Attendant Consoles
- M3000 telephones, and
- Meridian Digital Telephones equipped with digit display or display screens.

If System Park DNs are not defined for the customer, the default access ID is the DN of the telephone where the call was parked. An attendant must press the Park key and enter a DN if System Park DNs are not defined.

Park the call, then page the person called. The person called then picks up the call directly or through the attendant. Call Park also enables the telephone that originally receives the call to park it so that another telephone can retrieve it later. The telephone placing the call in Park is free to make or answer other calls.

Calls can be parked from telephones or Attendant Consoles with the Park key/lamp pair or Special Prefix (SPRE) code. Parked calls not retrieved within a specified time (30 to 240 seconds) are recalled to the telephone that parked it. Music for parked calls can be provided if Music (MUS) package 44 is installed.

If a call is parked on a System Park DN, it is recalled to the attendant who parked it if the parking attendant is idle. If the parking attendant is busy, the call is presented to any idle attendant.

If a call is parked on a telephone DN, the recall is placed in the attendant queue and presented to any available attendant. In all cases, parked calls recalled to the attendant appear on the Recall Incoming Call Identification (ICI) key, if defined.

The Park DN of the most recently parked call can be redisplayed on Meridian 1 proprietary telephones equipped with displays, a Park key, and a Display key. This is done by pressing the Display key, then the Park key. The attendant can display the last call parked by pressing the Park key when no loop key is active.

Operating parameters

Call Park is not available for calls on Dial Intercom keys or for calls on analog (500/2500 type) telephones designated as Dial Intercom telephones.

Call Park is not permitted when Privacy Release or Conference is active.

Calls parked from Meridian 1 proprietary telephones and analog (500/2500 type) telephones are recalled to the telephone that parked the call.

When a Multiple Appearance Single Call telephone mix (the same DN appears on Meridian 1 proprietary telephones, and single-line telephones) is parked, other appearances are not automatically bridged to the parked call when going off hook. The call can be retrieved by another Multiple Appearance DN (MADN) telephone only by dialing the Call Park retrieval code and the DN.

Remote access (for example, Centralized Attendant Service or Direct Inward System Access) for parked parties is not permitted.

Private lines, attendant DNs, Automatic Call Distribution (ACD), and Direct Inward System Access (DISA) DNs are not valid park numbers.

Trunks without disconnect supervision cannot be parked.

Parked calls are not retained during initialization or SYSLOAD.

Parked calls cannot be accessed with the Automatic Call Distribution (ACD) In-calls key. If parked access from Automatic Call Distribution (ACD) positions is required, a DN key must be provided.

A parked call recall cannot be placed on hold by the attendant.

A call transferred to the attendant by the Conference key on a Meridian 1 proprietary telephone cannot be parked by the attendant. A call transferred to the attendant by the Transfer key on a Meridian 1 proprietary telephone can be parked by the attendant.

Feature interactions

AC15 Recall: Transfer from Meridian 1

If Party Z parks the call initiated by Party X (an external caller), then the AC15 Recall: Transfer cannot be used to call Party Y. Party Z can neither park, selectively, one member of a split trunk nor park a whole split trunk. This avoids a recall to an attendant on the recall originating node that would not be able to send a recall to toggle from one party to another.

AC15 Recall: Transfer from Norstar

Remote access to call park from AC15 TIE trunks is not permitted. It is not possible to park an AC15 trunk if it has a call on hold. When an AC15 trunk is parked, it is not allowed to initiate a consultation call.

Access Restrictions

A call can be parked on any DN, regardless of its Class of Service. Access to a parked call is governed by the same Class of Service restrictions for normal trunk-to-telephone call processing. Table 32 details the restrictions. These restrictions can be overridden with the Authorization Code.

Table 32
Accessing telephone Class of Service

Parked call type	Accessing telephone Class of Service		
	FRE	FR1	FR2
Telephone	allowed	allowed	allowed
CO/FX/WATS	denied	denied	denied
DID Trunk	denied	denied	denied
TIE trunk	allowed	allowed	denied

Advice of Charge for EuroISDN

When a set parks a call charged with Advice of Charge, the calling party continues to be charged until the call is answered by another set.

Attendant Blocking of Directory Number

It is not possible to park an Attendant Blocking of DN call. If a Call Park call recalls to a blocked DN, the recall will be treated as if the DN is in a ringing state.

Attendant Break-In

The attendant cannot break in to a parked call.

Automatic Call Distribution

Calls parked by Automatic Call Distribution (ACD) agents are recalled to the ACD DN queue and presented to any available agent.

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Call Park (PRK) key is ignored.

Attendant Console - M1250/M2250

The Call Park access code and the Park DN are displayed for parked call recalls.

Attendant Console - QCW4

When a parked call returns to the console, the console shows an attendant display (DLEN in LD 12) of eight digits with only the Special Prefix (SPRE) code and the Park DN when a parked call recalls to the console. (Press the Display Destination key twice for the Park DN.) An attendant display of 16 digits shows the SPRE, the Call Park access code, and the Park DN for parked call recalls.

Autodial
Speed Call

Autodial and Speed Calls can be programmed to park calls or access parked calls.

Automatic Timed Reminders

A Call Park recall to an attendant appears on the Recall Incoming Call Indicator.

Busy Lamp Field

A busy lamp field can be equipped to display the status of System Park DNs.

Call Detail Recording (CDR)

Call Detail Recording (CDR) records for Call Park are similar to the start and end records generated when a call is transferred or terminated. When a call is parked, a Call Detail Recording (CDR) start record is generated if one has not already been generated by another feature. A CDR record is not generated when the parked call is accessed. A CDR end record is generated when the trunk call is terminated or when a parked call disconnects.

Call Detail Recording on Redirected Incoming Calls

There is no interaction with Call Detail Recording on Redirected Incoming Calls, as there is no “N” record generated in a Call Park scenario.

Call Forward

A recalled parked call to telephones with Call Forward, Call Forward Busy, or Call Forward No Answer (CFNA) is not forwarded.

Call Page Network Wide

A station set or Attendant Console that parks an external Call Page Network Wide (PAGENET) uncontrolled call is not blocked. However, an external PAGENET controlled call is blocked.

Call Park on Unsupervised Trunks

A 14-second Disconnect Timer applies to parked calls on all trunks on the route. All answered calls in the parked state will be disconnected if left in that state for an extended period.

Call Party Name Display

Upon valid operation of the Park key, or dial-access if used, Call Party Name Display (CPND) shows the SPRE code and the Park Access ID. Because the Park Access Code is displayed, no CPND name is displayed. The only time that the digit display shows the actual DN of the parked party is when the parked party has been retrieved, put on hold, and then retrieved from hold.

Call Pickup

An analog (500/2500 type) telephone user on a call can pick up a call by parking the existing call, then activating the Call Pickup feature.

Call Pickup Network Wide

The Call Pickup Network Wide feature cannot be used to pick up parked calls. A recall of a parked call can be picked up, in which case the call is unparked and answered by the requesting party.

Call Transfer
Conference

A parked call can be accessed after Call Transfer or Conference is activated.

Call Waiting

A recall of a parked call is not presented in the Call Waiting mode. If an internal telephone is in the parked state, Call Waiting to that telephone is not provided.

Centralized Attendant Service

Call Park is limited to the local system for systems equipped with Centralized Attendant Service. Call Park cannot be accessed from release-link trunks.

China – Attendant Monitor

If a DN being monitored becomes parked by another party, the Attendant Monitor feature is deactivated.

Conference

A parked call can be accessed after Conference is activated

Console Presentation Group Level Services

If the attendant who parked a call on the System Park DN is busy when that call is recalled and the parking attendant does not belong to the same Console Presentation Group (CPG) specified for the tenant of the calling station, the parked call is presented to an idle attendant in the same CPG specified for the calling station. If no attendant in that CPG is available to receive the recall, the parked call is queued until one of the attendants in the CPG becomes idle.

Tenant access checking between the set (A) who picks up a parked call and the party (B) who parked the call, is enforced as follows:

- If B is a set, tenant-to-tenant access must be allowed between A and B.
- If B is an attendant, A and B must belong to the same CPG for tenant-to-tenant access.
- If access is denied, set A (who intends to pick up the access-denied parked call) receives a blocking tone.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Attempts to intrude into a parked call receive Executive Intrusion Denied treatment.

Display of Calling Party Denied

When the Call Park timer expires on a parked call, a set's display reflects the Directory Number the call is parked against. The display does not include the name and DN of the calling party. When a parked call is retrieved by another set, display information is based on the DPD Class of Service of the individual sets.

Do Not Disturb

Calls can be parked on telephone DNs that are in the Do Not Disturb mode (DND). Telephones in the DND mode can park a call or access a parked call. Recall of a parked call to a DND telephone is recalled to the attendant.

Generic XFCOT Software Support

Call Park feature allows an attendant or telephone user to place a call in parked state (connected to a parked DN) where it can be retrieved by any attendant console or station set. If the call is not retrieved after a customer-defined time, the call is recalled to the telephone user who parked it.

Call Park is allowed on disconnect-supervised or unsupervised IPE loopstart Central Office trunks. If a caller on an unsupervised loopstart trunk disconnects while the call is in parked state is detected when the parked call is recalled or answered.

Caller disconnection during park state is detected by a disconnect supervised loopstart trunk on an XFCOT card. The disconnected caller is then dropped from the parked DN.

Group Call

Call Park cannot be applied on a Group Call.

Held Call Clearing

A call put on hold during a Call Park is not cleared by an on-hook action on that set.

Hot Line

Analog (500/2500 type) Hot Line telephones with EHTA and XFA Class of Service are allowed to park calls using the established Call Park procedures. Once a call is parked on an analog (500/2500 type) Hot Line telephone and the telephone is placed on hook, it cannot be unparked. Parked calls will recall to the parking telephone after the Call Park timeout. Two-way Meridian 1 proprietary telephone Hot Line stations that are equipped with a Call Park key/lamp pair are allowed to park calls in the normal fashion. As with analog (500/2500 type) telephones, a call parked from a Hot Line key cannot be picked up using the same key.

In-Band Automatic Number Identification

If an agent parks an In-Band ANI call and it times out and recalls the agent, the ANI number is not displayed.

INIT ACD Queue Call Restore

Parked calls are restored by ACDR as new incoming calls to the ACD DN.

Intercept Computer Dial from Directory - Pre-dial Operation

An attendant can park a call in the following manner:

- Press the Call Park key on the Attendant Console.
- Dial a DN from the Intercept Computer.

Terminate Call Park operation by pressing the Release key.

Make Set Busy

Recall of a parked call to a telephone in the Make Set Busy mode is intercepted by the attendant.

Multi-Tenant Service

If the attendant who parked a call on the System Park DN is busy when that call is recalled and the parking attendant does not belong to the same Console Presentation Group (CPG) specified for the tenant of the calling station, the parked call is presented to an idle attendant in the same CPG specified for the calling station. If no attendant in that CPG is available to receive the recall, the parked call is queued until one of the attendants in the CPG becomes idle.

Tenant access checking between the set (A) who picks up a parked call and the party (B) who parked the call, is enforced as follows:

- If B is a set, tenant-to-tenant access must be allowed between A and B.
- If B is an attendant, A and B must belong to the same CPG for tenant-to-tenant access.
- If access is denied, set A (who intends to pick up the access-denied parked call) receives a blocking tone.

Music

When a call is parked, music is not heard. When a trunk is parked, music plays if music is enabled for the route.

Network Intercom
Private Line Service

Hot Type I and Private Line Service calls cannot be parked.

Periodic Pulse Metering

When a metered call is parked from one station to another, the controlling station is charged until the call is answered.

Privacy Override

Calls in a Privacy Override conference state cannot be parked.

Privacy Release

When a call from a Meridian 1 proprietary telephone is parked, that telephone cannot activate Privacy Release. For example, Party A calls Party B. Party B parks the call. Party A cannot activate Privacy Release.

Recall After Parking

This enhancement to Call Park causes a parked call to be recalled to the attendant or night DN if the attendant is in Night Service, rather than to the parking telephone, if not answered within a customer-defined period of time (two-minute maximum). The call may be external or internal.

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the night DN. If the night DN is busy, the call is queued if it is an external call.

Traffic measurements

TFC007 is included for Call Park. It provides traffic measurements for the following:

- system park usage
- system park overflow
- telephone park usage
- park access
- park recall
- average waiting time

Feature packaging

Call Park (CPRK) is package 33 and has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable or disable Call Park.

2 LD 50 – Add/change or print Call Park. This overlay must be defined for Call Park operation.

3 LD 10 – Allow or deny access to Call Park for analog (500/2500 type) telephones.

4 LD 11 – Add or change a Call Park key on Meridian 1 proprietary telephones.

5 LD 12 – Add or change a Call Park key on Attendant Consoles.

LD 15 – Enable or disable Call Park.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	CPA	Enable Call Park.

LD 50 – Add/change or print Call Park. This overlay must be defined for Call Park operation.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CPK	Call Park data block.
CUST	xx	Customer number, as defined in LD 15

CPTM	30-(45)-240 30-(45)-480	Call Park Timer (in seconds). Call Park recall time (in seconds) if CPRK package 33 is equipped. The amount of time a call is held in the parked state before recalling the parking set or the attendant.
SPDN	(0)-50 xxxx	Number of contiguous System Park DNs and the first System Park DN. The default 0 (zero) disables System Park DN capability, but allows Telephone Park DNs. If the DN Expansion package is equipped, the System Park DN can have up to seven digits.
MURT	0-511 0-127	Music route number for parked calls. For Large Systems For Small Systems and Succession 1000 systems

LD 10 – Allow or deny access to Call Park for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	XFA	Allow access to Call Park.

LD 11 – Add or change a Call Park key on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx PRK	Add a Call Park key (key number must be 17 for the M2317 and 31 for the M3000).

LD 12 – Add or change a Call Park key on Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	1250 2250	Attendant Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx PRK	Add a Call Park key (key number can be 00-19 on the M2250).

Feature operation

To park a call with the Park key:

1 Press **Park** twice.
If there is a System Park extension, the call is parked on it. Otherwise, it is parked on your extension.

To park a call on an extension other than the System Park extension, follow these steps:

1 Press **Park**.

2 Enter the extension number.

3 Press **Park** again.

To park a call using SPRE codes, follow these steps:

1 Press **Transfer** or **Conference**.

2 Dial SPRE 71.

You can dial an extension number to park the call, or you can use the System Park extension, chosen automatically. It shows on your telephone's display, if equipped.

3 Press **Transfer** or **Conference** again.

To retrieve a parked call, follow these steps:

1 Select a free extension.

2 Dial SPRE 72.

3 Dial the extension where the call is parked.

Call Park on Unsupervised Trunks

Contents

This section contains information on the following topics:

Feature description 763
Operating parameters 764
Feature interactions 764
Feature packaging 765
Feature implementation 765
Feature operation 765

Feature description

This enhancement to the Call Park feature allows Central Office (CO), FEX, and Wide Area Telephone Service (WATS) trunks, without disconnect supervision, to be call-parked. All other trunk types without disconnect supervision cannot be parked. The Disconnect Timer (DCTI) is used to prevent phantom calls from ringing beyond the set time. Answered calls in the held, parked, camped-on, or ringing state are disconnected when the DCTI times-out.

This enhancement also allows Direct Inward System Access (DISA) on CO, FEX, and WATS trunks without disconnect supervision. DISA on unsupervised trunks does not intercept to the attendant, but is subject to Timed Forced Disconnect Timer, which prevents the CO trunk from being seized if the far end hangs up.

Operating parameters

The Disconnect Timer apples not only to Call Park but also to all trunks on the route. All answered calls in the held, parked, ringing, or Camp-On states will be disconnected if left in that state for an extended period (this even applies to calls in a call waiting queue type).

Feature interactions

Attendant Calls Waiting Indication

If all the attendants are busy and a Call Park Recall occurs, the recall is placed in the calls waiting queue. If the recalled station is busy when the recall occurs, the Disconnect Timer (DCTI) temporarily suspends timing until the recall is presented. After the recall is presented, the Disconnect Timer continues timing for the remainder of the period.

Automatic Call Distribution (ACD)

If all the ACD agents are busy and a Call Park Recall occurs, the recall is placed in the ACD DN queue.

Call Hold, Deluxe
Call Hold, Permanent

A Disconnect Timer applies to held calls on all trunks on the route. All answered calls in the held state will be disconnected if left in that state for an extended period.

Call Park

A 14-second Disconnect Timer applies to parked calls on all trunks on the route. All answered calls in the parked state will be disconnected if left in that state for an extended period.

Camp-On

A Disconnect Timer applies to camped-on calls on all trunks on the route. All answered calls in the camped-on state will be disconnected if left in that state for an extended period.

Feature packaging

Call Park on Unsupervised Trunks is included in Direct Inward System Access (DISA) package 22.

Feature implementation

LD 16 – Set the disconnect timer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
...		
DCTI	(0)-511	Time, in seconds, that an extension is allowed to ring or be on hold or Call Park before the trunk is disconnected. 0, the default, or <CR> means that the condition goes on indefinitely. Respond with a value equal to the number of seconds a set is to ring after recall, plus the value of the Call Park Recall Timer (which is defined in LD 50 in response to the CPTM prompt). The value stored, which will be the closest lower multiple of four, is echoed back upon entry.

Feature operation

Feature operation for Call Park on Unsupervised Trunks is the same as that for Call Park.

Call Party Name Display

Contents

This section contains information on the following topics:

Feature description 767
Operating parameters 771
Feature interactions 772
Feature packaging 778
Feature implementation 778
Feature operation 785

Feature description

Call Party Name Display (CPND) identifies the calling or called number in addition to the DN. The identifier (for example, the name) associated with a DN on telephones with an alphanumeric display is defined in LD 95.

Whenever the calling party's DN displays on the terminating telephone, the calling party's name also appears. Likewise, on an internal call, the called party's name is appended to the displayed DN on the originator's telephone, as soon as a valid DN is completely dialed.

CPND displays the DN and name of the originally dialed party for redirected calls. A Class of Service, DNDA/DNDD (Dialed Name Display Allowed or Denied), is assigned on a per-telephone basis. The terminating telephone must have DNDA to display the name of the originally dialed party.

The M1250/M2250 Attendant Console can extend a call to a DN requested by a calling party. The CPND enhancement enables the M1250 Attendant Console to display the incoming call information on one line and the outgoing call information on the next line when extending an incoming call.

Multi-Language CPND displays the party's name in Roman/English or Katakana (Japanese alphabet) characters on Meridian modular telephones. The names are stored in the database under each character set and the language is specified with the Meridian modular program keys.

Two languages can be stored in the database for any given name. For this enhancement to work fully, both telephones involved must have the same name in the same languages. (For example, John Smith calls Anne Jones. Both John and Anne must have Katakana in their database for the name to appear in Katakana characters. If John has Katakana enabled, but Anne does not, Anne sees the English version.)

Entering Katakana, or any other non-ASCII Roman characters, requires a system terminal that supports eight-bit, no-parity Input/Output.

The maintenance terminal must support ISO 8859-1 Latin 1 for the Roman character mode.

Call Party Name Display assignment

A CPND name string can be assigned to internal DNs associated with any of the following:

- analog (500/2500 type) telephones
- Trunk access codes
- Attendant DNs
- Automatic Call Distribution (ACD) DNs
- Dial Intercom Group member numbers

As a customer option for multiple appearance DNs (MADNs), the assigned CPND name can be linked with its member telephone's designator (DES field in the TN block) to further identify the party of a shared DN.

Call Party Name Display composition

A CPND name is the name used to identify a DN, entered in ASCII alphanumeric character format. The maximum CPND length is the smaller of two values: the maximum length configured in LD 95 or 27 characters, including spaces and special characters.

The ASCII characters supported are A-Z, 0-9, space, Hex 20-127, and the following special characters:

" ( ) - # ,

The NAME prompt in LDs 10, 11, and 95 accepts first name, a comma as a separator, and last name (such as Mary,Smith). CPND also supports names using a space separator (such as Mary Smith), treating the entire name string as the first name. See Table 33 for examples.

Table 33
Response formats for CPND NAME prompt

Entered Data	Displayed Result
Sue Smith <CR>	Sue Smith
Sue,Smith <CR>	Sue Smith
Sue <CR> Sue, <CR>	Sue (Trailing comma is ignored.)
Sue,Smith, Dept. 410 <CR>	Sue Smith, Dept. 410
Sue Smith, Joe Brown <CR>	Sue Smith, Joe Brown

The default is to accept the names as entered, replacing the comma with a space. Hence, a value entered as Mary,Smith displays as Mary Smith.

> ***Note:*** Do not enter leading spaces. LD 95 ignores them. When CPND information is printed (using LD 10/11 or LD 20), the printout reflects what is in the database, not what appears on the telephone display.

In addition to the caller's name, a reason field can be provided to indicate the cause of a redirection. This is a customer option and the actual mnemonics are service changeable. The following call redirections have a reason displayed:

- Call Forward All Calls

- Call Forward No Answer
- Hunting/Call Forward Busy
- Call Transfer with Network Call Redirection
- Attendant Alternative Answering
- Call Pickup

Display Devices and Capabilities

The M3000 Touchphone has a display line of 35 characters, 27 available for displaying DN-related information.

The M2317 has a display line of 40 characters, 33 available for displaying DN-related information.

If there are more characters than the telephone's display allows, the system deletes letters to make the name fit.

The M1250 and M2250 Attendant Consoles are equipped with four lines of LCD alphanumeric display. Each line has 40 characters, and lines 2 and 3 are used to display DN-related information. If the number of characters displayed is more than 40, an arrow appears in the upper right corner of the display. The arrow alerts the user that more information can be retrieved using the scrolling keys. For complete information, refer to the *M1250/M2250 Attendant Console User Guide*

The call type, originating or terminating telephone, and the Class of Service all affect the display and CPND information. Three Classes of Service are associated with the display function. CPND conforms to whichever Class of Service is configured for the telephone.

- Automatic Digit Display (ADD)
- Digit Display Selection (DDS)
- Touchphone Digit Display (TDD)

No user interaction is required to display information on the call. On the M2317 telephone, however, the user can press the SAVE # softkey to save the name and number of the calling party. This applies to all outgoing and answered incoming calls.

Operating parameters

CPND is not displayed if a live call is not involved (for example, while programming a Speed Call key).

Attendant Administration does not support the entry of CPND class marks for digital telephones.

CPND is not displayed on the calling telephone while making an outgoing trunk call.

CPND is not supported on data calls.

CPND is not available on QCW Attendant Consoles.

CPND applies only to redirected calls on M2008, M2016, M2216, M2616, M3000, and M2317 telephones.

For M2008, M2016, M2216, M2616, M3000, and M2317 telephones, CPND is provided on a per-telephone basis, depending on the Class of Service.

DNDA (Dialed Name Display Allowed) and NDD (No Digit Display) Class of Service are mutually exclusive.

Multi-Language CPND operates on Meridian modular telephones only.

An individual DN can have Roman/English, or Katakana, or both programmed in the database if MLIO is equipped.

If the call destination is a trunk or a telephone type other than Meridian modular, the name is translated into the ASCII equivalent.

Multi-Language CPND applies to DNs on local switches only. CPND for Integrated Services Digital Network (ISDN) calls is displayed in English only.

The CPND feature uses JIS X 0201-1976, the RCode for Information Interchanges, also known as JIS-Roman, which specifies the upper- and lower- case letters, numbers, punctuation and symbols, and Katakana.

Feature interactions

AC15 Recall: Timed Reminder Recall

When the AC15 recall is presented to an attendant or a set with a display, the source and destination names are shown beside the DNs or the ACODs.

ACD Routing by DNIS

When an incoming trunk call from a route with Routing by DNIS is presented to a display telephone, the identification digits follow the regular trunk access code and member number. It precedes the CPND name for the DNIS DN.

Attendant Recall

Attendant Recall using the Attendant Recall key or a switchhook flash results in both source and destination information being displayed. No redirection reason is displayed, however. In this type of recall, the party that pressed the Attendant Recall key or switchhook is the destination party.

Attendant Recall using Call Transfer or Conference displays the recalling party's DN and CPND information on the attendant's source line. No redirection reason is displayed. If the recall is done with the Transfer key the third party's DN and CPND information are displayed on the source line when the transfer is complete.

Attendant Recall with Splitting

For the M1250 and M2250 Attendant Console, M2317, M3000 digital sets, and Meridian Modular sets the appropriate DN and calling party's name will be correctly shown on the digit display when the attendant presses either the Exclude Source or the Exclude Destination key.

Autodial
Speed Call

No name information displays during the programming of Autodial and Speed Call numbers.

Automatic Call Distribution (ACD) Dialed Number Identification Services (DNIS)

If an incoming trunk call from a route with DNIS is presented to a display telephone, the identification digits follow the regular trunk access code and member number. It precedes the CPND name for the DNIS incoming trunk group.

Automatic Wake Up

All display information associated with Automatic Wake Up (AWU) programming is directed to line three of the display. Names are appended to DNs appearing on line three if they are different from those on line two, or if no DN appears on line two. There is no DN information on line two if the attendant has initiated the AWU process while not on an active call. No DES information is appended, since AWU operates on a DN basis.

Call Hold, Deluxe

When a call is put on hold, the holding telephone's display clears. The held telephone's display does not change. When the telephone reestablishes the call, the display returns the original DN and name.

Call Park

Upon valid operation of the Park key, or dial-access if used, CPND shows the SPRE code and the Park Access ID. Because the Park Access Code is displayed, no CPND name is displayed. The only time that the digit display shows the actual DN of the parked party is when the parked party has been retrieved, put on hold, and then retrieved from hold.

Call Pickup

For Call Pickup, CPND applies when the call is answered.

Call Pickup Network Wide

Network Call Party Name Display information will be exchanged during Call Pickup Network Wide calls if the sets involved in the call would normally exchange the information for calls over the routes that have been used for the original call and the call pickup. Conversely, if Network Called Party Name Display would not operate for a normal call from the originating party to the terminating party, the service will not be supported when Call Pickup Network Wide is involved.

Call Transfer

When the Transfer key is pressed during an active call, the display clears. The call is in a held state. The DN and name of the transferred telephone appear on the display when the DN is dialed. When the transfer is complete, the transferring telephone's display clears because the telephone is now disconnected. The transferred telephone's display changes to show the name of the newly connected party.

Calling Party Privacy

In current operations, if the International Supplementary Features (SUPP) package 131 is not equipped in the system, an incoming ISDN call with the Call Party Name Display (CPND) Indicator field set to "Presentation Denied" still displays the Calling Party Name. If package 131 is equipped in the system, the current operations will inhibit the Calling Party Name for an incoming ISDN call with the CPND Indicator field set to "Presentation Denied".

The CPP feature will inhibit the display of the Calling Party Name for an incoming ISDN call with the CPND Indicator field set to "Presentation Denied" if package 131 is not equipped.

Centralized Attendant Service (CAS)

When an attendant in the CAS mode extends a call to a remote station, the display shows only the source line.

Conference

When pressed during an active call, or to set up a conference, the Conference, Connect, or Join Parties key clears the display. The telephones involved in the conference have blank displays. If the conference returns to a two-way only call, each telephone displays the DN and name of the other telephone.

Dial Intercom

The display on telephones connected by Dial Intercom shows the group member's DIG number plus CPND information.

Dialed Number Identification Service

If an incoming trunk call from a route with Dialed Number Identification Service (DNIS) is presented to a display telephone, the identification digits follow the regular trunk access code and member number. It precedes the CPND name for the DNIS incoming trunk group.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Call Party Name Display feature is supported in a DPNSS1 UDP network. Names can be associated with the access codes of the DPNSS1 UDP routes defined in LD 95.

Display Key

When pressed during a call, the Display key clears the display until pressed again. The original display reappears. When the telephone is inactive and the DSP key is pressed, followed by a function key like Autodial, no CPND information is displayed.

End-to-End Signaling

When entered after a call is answered, EES digits are displayed immediately following the CPND name of the connected party. Leading DN digits and name characters may be shifted out of the display window.

ISDN

On incoming ISDN calls, the Calling Line ID number can be displayed instead of a DN on the source party line. CPND applies to telephones configured for ISDN when redirection is supported. CPND allows calls to redirect across a network with Network Call Redirection. The CPND is maintained through the redirection.

ISDN QSIG Name Display

Call Party Name Display and Calling Party Name Display Denied interact with ISDN QSIG Name Display, depending on the Name Display configuration in LD 16 for BRI or LD 17 for PRI. When a QSIG network is interacting with an MCDN network providing network capability ND3, both the MCDN and QSIG Name Display feature function on the same level.

Listed Directory Number

CPND is not supported for LDNs. If the LDN is an incoming trunk route, the CPND assigned to the route access code is displayed.

Manual Signaling (Buzz)

If the Signal key is pressed to buzz another telephone, no digit or name display appears on the telephone.

Meridian Hospitality Voice Services

The maximum length of a CPND name sent from the PMSI/Background Terminal (BGD) is 27 characters. When the full 27-character length is used, part of the CPND name may scroll off the screen. To avoid this problem, the PMSI/Background Terminal (BGD) software has been updated to strip from the screen all trailing blanks from the CPND name.

Meridian Mail Voice Mailbox Administration

There is significant interaction between the Meridian 1 Call Party Name Display (CPND) database and the Meridian Mail VMB database. These interactions are further described in the “Meridian Mail Voice Mailbox Administration” description in *Features and Services* (553-3001-306), Book 2.

Meridian 911

The Call Party Name Display feature can be used to configure and display the incoming 911 route name.

M2312 Digit Display

The calling party number can be displayed only when the call is active.

Network Intercom

Hot Type I calls display names the same as a normal call.

Hot Type I calls that become a normal call indicate on the originating station's display that the call is no longer a Hot Line call.

Override

When Overriding an established call, the displays of the other telephones show the DN and name of the overriding party.

Slow Answer Recall

Slow Answer Recall results in displays showing source and destination information. If a redirection occurs, the reason is displayed.

Telephones - M3000 Touchphone

Local Directory Translation CPND and the M3000 Touchphone DN-to-name translation are mutually exclusive. If CPND name display is allowed (CLS = CNDA), the DN-to-name translation must be disallowed.

Voice Call

The telephone originating a Voice Call displays the called DN's CPND. The called telephone shows the caller's DN and name on its display.

Feature packaging

Call Party Name Display (CPND) package 95 requires:

- Digit Display (DDSP) package 19
- M2000 Digital Sets (DSET) package 88
- M3000 Digital Sets (TSET) package 89 or
- M2317 Digital Sets (DLT2) package 91
- Aries Digital Sets (ARIE) package 170

Multi-Language CPND requires Multi-Language TTY Input/Output (MLIO) package 211.

If the designator field is to be used for multiple-appearance DNs, CPND requires:

- Office Data Administration System (ODAS) package 20

For Hotel/Motel applications configuring CPND, CPND requires:

- Background Terminal Facility (BGD) package 99
- Multi-Language TTY Input/Output (MLIO) package 211 to support eight-bit, no-parity system terminals

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 95 – Create the CPND data block.

2 LD 95 – Add names to the CPND data block.

3 LD 10 – Allow names to be assigned to analog (500/2500 type) telephones.

4 LD 11 – Allow names to display on M2008, M2016, M2216, M2616, M3000, and M2317 telephones.

5 LD 12 – Allow names to display on Attendant Consoles.

6 LD 95 – Open the CPND data block to change or remove entries.

7 LD 95 – Print information associated with entries in the CPND data block.

8 LD 10/11 – Add or change CPND name.

Note: Before name strings can be assigned to various telephones, the CPND data block must be created in LD 95. The number and size of CPND name strings is limited by available space in the Protected Data Store, so it is recommended that you initially use a small number for the maximum character length.

Enable CPND and add names to the CPND data block

LD 95 – Create the CPND data block.

Prompt	Response	Description
REQ	NEW	Create CPND database (or open existing database).
TYPE	CPND	CPND data block.
CUST	xx	Customer number, as defined in LD 15
CNFG	<CR>	Standalone memory.
MXLN	5-(17)-27	Maximum number of characters allowed in each name string. Once defined, this value can be changed only by removing the CPND data block and recreating it.
STAL	(NO) YES	Static allocation of name storage. Must be YES if Background Terminal is equipped, or whenever name strings change frequently.

- DFLN	5-MXLN	Average default character string length. Suggested default is 13 or the maximum length given to MXLN, whichever is less. Prompted if STAL = YES.
DES	(NO) YES	(Do not) allow designator for MADNs.
RESN	(NO) YES	(Do not) allow display of reason for redirecting calls.
- CFWD	aaaa (F)	Mnemonic for Call Forward All Calls display.
- CFNA	aaaa (N)	Mnemonic for Call Forward No Answer display.
- HUNT	aaaa (B)	Mnemonic for Hunt/Call Forward Busy display.
- PKUP	aaaa (P)	Mnemonic for Call Pickup display.
- XFER	xxxx (T)	Mnemonic for Call Transfer display for NCRD.
- AAA	aaaa (A)	Mnemonic for Attendant Alternative Answering.

LD 95 – Add names to the CPND data block.

Prompt	**Response**	**Description**
REQ	NEW	Open CPND data block to add new entries.
TYPE	NAME	Create a new name string.
LANG	(ROM) KAT <CR>	Store the name in Roman or Katakana. <CR> stores the name in English.
CUST	xx	Customer number, as defined in LD 15
DIG	0-2045 0-99	Dial Intercom Group number and member number. Each time a name string is assigned to a Dial Intercom Group member, the DIG prompt repeats, until a carriage return is entered to go to the DN prompt.
	<CR>	Bypass Dial Intercom Group and go to the DN prompt to assign names on a DN basis.
- NAME	aaaa bbbb	CPND name string; maximum of 27 characters.
- XPLN	xx	Expected Length. Range must be between the Input Name length and the MXLN, or it defaults to DFLN.
	<CR>	Set XPLN to average default character string length (DFLN) or the actual length (NAME), whichever is longer.
DN	xxx...x	DN to which name string is linked.

- NAME	aaaa bbbb	CPND name string; maximum of 27 characters.
- XPLN	xx	Expected Length. Range must be between the Input Name length and the MXLN, or it defaults to DFLN.
	<CR>	Set XPLN to average default character string length (DFLN) or the actual length (NAME), whichever is longer.
DCNO	xxx	IDC conversion table number (0-254).
IDC	nnn	Existing complete or partial IDC number. Prompted only when DCNO is valid.
NAME	aaaa bbbb	CPND name string; maximum of 27 characters.

LD 10 – Allow names to be assigned to analog (500/2500 type) telephones.

Prompt	**Response**	**Description**
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FTR	CPND	Allow CPND name assignment on this telephone.

LD 11 – Allow names to display on M2008, M2016, M2216, M2616, M3000, and M2317 telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = 2008, 2016, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CNDD) CNDA (DNDD) DNDA	(Deny) allow display of CPND entries. (Deny) allow display of CPND originally dialed entries.

LD 12 – Allow names to display on Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CPND	(CNDD) CNDA	(Deny) allow CPND name assignment.
DNDI	(DNDD) DNDA	(Deny) allow display of originally dialed CPND entries.

Change or remove names in the CPND data block

LD 95 – Open the CPND data block to change or remove entries.

Prompt	Response	Description
REQ	CHG OUT	Change, or remove an existing entry.
TYPE	NAME	Change, or remove an existing CPND name string.
CUST	xx	Customer number, as defined in LD 15
LANG	ROM KAT ALL	Change or remove the name in Roman or Katakana. ALL is used to remove all names stored for the DIG.
DIG	0-2045 0-99 ALL <CR>	Dial Intercom Group number and member number. Each time a name string is assigned to or removed from a Dial Intercom Group member, the DIG prompt repeats, until a carriage return is entered to go to the DN prompt. ALL removes all entries for that DIG. <CR> bypasses DIG and goes to the DN.
- NAME	aaaa bbbb	CPND name string for this DIG; maximum of 27 characters.
	<CR>	Leave this entry unchanged.
DN	xxx...x	DN of name string being changed or removed.
	ALL	Remove all DN-defined entries.
	<CR>	Return to REQ prompt.
- NAME	aaaa bbbb	CPND name string; maximum of 27 characters.
DCNO	xxx	IDC conversion table number (0-254).
- IDC	nnn	Existing complete or partial IDC number. Prompted only when DCNO is valid.
NAME	aaaa bbbb	CPND name string; maximum of 27 characters.

Print entries from the CPND data block

LD 95 – Print information associated with entries in the CPND data block.

Prompt	Response	Description
REQ	PRT	Print entries in the CPND data block.
TYPE	NAME	CPND name strings.
CUST	xx	Customer number, as defined in LD 15
LANG	ROM KAT	Print names in Roman or Katakana.
PAGE	(NO) YES	Page headers and page numbers for multiple DNs and DIGs.
DIG	ALL	Print information on all entries defined by Dial Intercom Groups.
	0-2045 0-99	Dial Intercom Group and member number. The DIG prompt repeats until a carriage return is entered.
	<CR>	Bypass Dial Intercom Group and go to the DN prompt to print information on a DN basis.
DN	ALL	Print information on all DN entries.
	xxxx	DN to print information from. DN prompt repeats until a carriage return is entered.
	<CR>	Return to REQ prompt.
DCNO	xxx	IDC conversion table number (0-254).
- IDC	nnn	Existing complete or partial IDC number. Prompted only when DCNO is valid.
	ALL	All names defined are printed.
SHRT	(NO) YES	Short form. Prints one IDC per single line. Prints several IDCs on single line.

Add or change CPND name entry for a telephone

LD 10/11 – Add or change CPND name.

Prompt	Response	Description
REQ:	NEW CHG	Add or change CPND name information.
TYPE:	aaaa	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
CPND	NEW CHG OUT	Add, change, or remove the CPND information.
CPND_LANG	(ROM) KAT	Use Roman or Katakana characters.
NAME	aaaa bbbb	CPND name; maximum of 27 characters.
XPLN	xx	Expected name length.
DISPLAY_FMT	(FIRST) LAST	First name; Last name (John Doe). Last name; First name (Doe John).

Feature operation

No specific operating procedures are required to use this feature.

Call Pickup

Contents

This section contains information on the following topics:

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Feature description

Call Pickup allows telephones to be arranged in groups consisting of any combination of analog (500/2500 type) telephones, and Meridian 1 proprietary telephones.

Telephones can be specified as Call Pickup allowed or Call Pickup denied. If the telephone's Class of Service is Call Pickup allowed, the user can answer calls made to any telephone within the Call Pickup group. If the telephone's Class of Service is Call Pickup denied, but the telephone is assigned to a Call Pickup group, the user cannot answer calls directed to other telephones. Calls to the denied telephone, however, can be answered by other members of the group.

Meridian 1 proprietary telephones can dial-access this feature, or be equipped with a Call Pickup key. An associated lamp is not required.

Central Office (CO) Trunk Priority, provides CO trunk calls priority over other calls within the distinctive ringing and normal ringing queues. If the CO Trunk Priority is implemented, calls are answered in the following order:

- Distinctive Ringing Queue CO call (Priority 1)
- Distinctive Ringing Queue non-CO call (Priority 3)
- Normal Ringing Queue CO call (Priority 2)
- Normal Ringing Queue non-CO call (Priority 4)

Operating parameters

The number of Call Pickup groups is 4095. The number of members assigned to each group is unlimited, depending on available system memory.

Feature interactions

Advice of Charge for EuroISDN

Calls charged with Advice of Charge that are either transferred, extended or redirected to another set using Call Pickup are charged against the last set that answers the call and the controlling set releases.

Attendant Alternative Answering

The Attendant Alternative Answering (AAA) DN can be assigned to a Call Pickup group to allow members of the same group to answer the call.

Attendant Overflow Position

An Attendant Overflow Position Call presented to the AOP DN can be picked up by any station belonging to the same Call Pickup Group.

Automatic Call Distribution

Automatic Call Distribution (ACD) DNs are not supported by Call Pickup.

Call Park

An analog (500/2500 type) telephone user on a call can pick up a call by parking the existing call, then activating the Call Pickup feature.

Calling Party Privacy

If an incoming trunk call with the Privacy Indicator is picked up locally, the display of the calling Party Number and Name are not displayed on the terminating set.

Call Detail Recording on Redirected Incoming Calls

When an incoming call is picked up, the answering set is identified in the Terminating ID. This operation remains the same.

Call Pickup, Directed

Call Pickup can be assigned to a telephone independent of Directed Call Pickup (DCP).

Conference

This feature cannot be activated during a conference call. Meridian 1 proprietary telephones can activate Call Pickup if an idle Directory Number (DN) key is available. The conference call must be put on hold before pressing the idle DN key to pick up the call.

Dial Intercom

Call Pickup may be used by Meridian 1 proprietary telephones if the telephones are all in the same Dial Intercom Group (DIG) and Call Pickup Group and the ring option is specified for the DIG

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

Call Pickup is supported in a DPNSS1 UDP network.

Display of Calling Party Denied

When a call is picked up from another set, the terminating set's display is in accordance with the Class of Service of the dialed and calling sets. The calling party's display includes the dialed DN, the terminating DN and the name of the terminated set. However, if the terminating set has Digit Display Denied (DDGD), then both the dialed and terminating sets' DNs are blocked from the calling party's display. The same occurs when Digit Display Allowed (DDGA) is configured on the terminating set. Both the dialed and terminating sets' DNs are displayed on the calling party's set, regardless of the Class of Service of the dialed set.

Flexible Feature Codes

Flexible Feature Codes are not supported on a Meridian 1 proprietary telephone during an attempt to pick up a Dial Intercom ringing call.

Group Call

This feature can be used to answer a Group Call if it is activated by a valid telephone in the same Call Pickup group, or by using Directory Number (DN) Pickup or Group Pickup.

Hot Line

Telephones with two-way Hot Line keys, and analog (500/2500 type) Hot Line telephones, can be assigned to pickup groups. Incoming Hot Line calls may be picked up by group members. To prevent someone from picking up a Hot Line call, do not put the Hot Line user into a Call Pickup group.

ISDN QSIG Name Display

An incoming QSIG call with name display presentation allowed has name information displayed on the set that picks up the call. If the incoming QSIG call has presentation denied, the calling party's name is not displayed on the set picking up the incoming call.

Multi-Party Operations

Analog (500/2500 type) telephones with PUA and TSA Class of Service can pick up a call only if they are not involved in another call. After picking up a call, the user can form a Consultation connection and dial Programmable Control Digits as normal.

Network Intercom

Hot Type I calls cannot be picked up. An attempt to pick up a Hot Type I call results in an overflow tone.

Periodic Pulse Metering

Metered calls transferred or extended from one station and answered at another station using the Call Pickup feature are charged against the station where the call is picked up as the controlling party disconnects.

Virtual Office Telephones - M3900 series only

The Call Pickup feature is not supported for Virtual Office Telephones. A Virtual Office Telephone cannot be a member of a call pickup group since the DPU and GPU keys cannot be configured.

The RNPG prompt is blocked in LD 11. Dialing a SPRE code for Call Pickup from a logged in Virtual Office Telephone results in overflow.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Implement CO Trunk Priority in the Customer Data Block.

2 LD 10 – Define Call Pickup group and Class of Service for analog (500/2500 type) telephones.

3 LD 11 – Define Call Pickup group, Class of Service, and Call Pickup key for Meridian 1 proprietary telephones.

LD 15 – Implement CO Trunk Priority in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(COX) COP	CO Trunk Priority for the Call Pickup feature. COX is no Priority.

LD 10 – Define Call Pickup group and Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
RNPG	0-4095	Call Pickup group number.
CLS	PUA	Allow Call Pickup.

LD 11 – Define Call Pickup group, Class of Service, and Call Pickup key for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
RNPG	0-4095	Call Pickup group number.
CLS	PUA	Allow Call Pickup.
KEY	xx RNP	Add a Call Pickup key.

Feature operation

To answer a call in your Call Pickup group from a Meridian 1 proprietary telephone, follow these steps:

1. Lift the handset, or press a DN key.
2. Press **Call Pickup** or dial SPRE + 3.

To answer a call in your Call Pickup group from an analog (500/2500 type) telephone, follow these steps:

1. Lift the handset.
2. Dial SPRE 3 or PURN FFC.
   You are connected to the caller.

   Note: If you are on a call when another call comes in for someone in your Call Pickup group, you must end, park, or transfer the existing call before you can answer the new call.

Call Pickup, Directed

Contents

This section contains information on the following topics:

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Feature description

Directed Call Pickup (DCP) allows a caller from one Call Pickup group to pick up a ringing call in another Call Pickup group. The ringing call is picked up by dialing either its Call Pickup Group number or the DN on which it is ringing.

Directed Call Pickup adds two new methods of Call Pickup to the existing Call Pickup feature:

- Group Pickup (GPU), and
- DN Pickup (DPU).

Group Pickup lets you pick up any ringing call in your own pickup group, or any pickup group in the system.

DN Pickup allows pickup of a call ringing on a specified DN. If a DN is not assigned to any group, it defaults to Group Zero (0). This prevents any other group from picking up that DN.

Both GPU and DPU can be activated using programmable keys or Special Prefix (SPRE) code dialing. Each pickup method can be assigned to a telephone independent of the others.

The dialed digits (DN or group number) are displayed on the Digit Display as dialed. Like the Call Pickup feature, the lamp is optional for the Call Pickup and Group Call Pickup keys. No second dial tone is given after the key is pressed, nor is it given after the SPRE code is dialed.

Operating parameters

Group 0 (zero) is not a valid group number. A telephone that is not part of any group is assigned by default to group 0 (zero).

Feature interactions

Automatic Call Distribution

Automatic Call Distribution DNs are not supported by Directed Call Pickup.

Call Pickup

Call Pickup can be assigned to a telephone independent of Directed Call Pickup (DCP).

Flexible Feature Code

Flexible Feature Codes are not supported on a Meridian 1 proprietary telephone during an attempt to pick up a Dial Intercom ringing call.

Multi-Party Operations – Three-Party Service

Users of analog (500/2500 type) telephones involved in a Three-Party Service call cannot pick up another call by dialing the SPRE code.

Feature packaging

Directed Call Pickup (DCP) is package 115 and has no feature package requirements.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Define the number of digits dialed for Call Pickup groups.
2. LD 10 – Configure analog (500/2500 type) telephones to allow DCP Class of Service.
3. LD 11 – Configure Meridian 1 proprietary telephones to allow Directed Call Pickup Class of Service.

LD 15 – Define the number of digits dialed for Call Pickup groups.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT		Options.
	(COX) COP	Central Office call No Priority for Ringing (default). Central Office call Priority for Ringing.
- PKND	(1)-4	Number of digits dialed for Group Pickup. Prompted only if DCP is equipped.

LD 10 – Configure analog (500/2500 type) telephones to allow DCP Class of Service.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
RNPG	(0)-4095	Ringing Number Pickup Group. 0 = no pickup group.
CLS	(GPUD) GPUA (DPUD) DPUA	(Deny) allow Group Pickup. (Deny) allow DN Pickup.

LD 11 – Configure Meridian 1 proprietary telephones to allow Directed Call Pickup Class of Service.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
RNPG	0-4095	Call Pickup Group. 0 = no pickup group.
CLS	(GPUD) GPUA (DPUD) DPUA	(Deny) allow Group Pickup. (Deny) allow DN Pickup.
KEY	xx GPU xx DPU	Group Pickup key (not available on M3000). DN Pickup key (not available on M3000).

Feature operation

To answer a call in another Call Pickup group from a Meridian 1 proprietary telephone, follow these steps:

1 Lift the handset.

2 Press **GRP Pickup** or dial SPRE + 94 or PUGR FFC.

3 Dial the pickup group number.

To answer a call on a specified DN from a Meridian 1 proprietary telephone:

1 Lift the handset.

2 Press **DN Pickup** or dial SPRE + 95 or PUDN FFC.

3 Dial the extension number.

To answer a call in another Call Pickup group from an analog (500/2500 type) telephone, follow these steps:

1 Lift the handset and dial SPRE + 94 or PUGR FFC.

2 Dial the pickup group number.

To answer a call on a specified DN from an analog (500/2500 type) telephone:

1 Lift the handset and dial SPRE + 95 or PUDN FFC.

2 Dial the extension number.

Call Redirection by Day

Contents

This section contains information on the following topics:

Feature description 801
Operating parameters 802
Feature interactions 802
Feature packaging 805
Feature implementation 805
Feature operation 809

Feature description

Call Redirection by Day (CRDAY) is an enhancement of the feature Call Redirection by Time of Day (CRTOD). The CRDAY feature allows you to automatically redirect incoming calls on specified days of the week and/or holidays. You can define the number of rings required before a call is redirected, and the Directory Number (DN) to which the call is redirected.

Existing parameters that apply to the operation and redirection of DNs of Call Forward No Answer (CFNA) and Hunting also apply to this feature. The CRDAY feature also uses the alternate DNs introduced by the CRTOD feature.

You can configure up to four alternate day lists, DAY0 – DAY3, for each customer. Each day list can contain one or more days of the week. You can also configure up to four holiday lists, HOLIDAY0 – HOLIDAY3, for each customer. Each holiday list can contain up to 20 dates.

You can assign one alternate day list and/or one alternate holiday list for each telephone. The maximum value of the year in a holiday date is 2104.

There is a class of service (CLS) for each of CRTOD, CRDAY, and Call Redirection by Holiday (CRHOL). Each CLS can be enabled or disabled separately and are independent of each other.

Operating parameters

CRDAY is not supported on Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) sets.

Feature interactions

Call Redirection by Time of Day

The CRTOD and CRDAY features can work together or separately. If all three classes of service are enabled, CRHOL takes precedence over CRDAY and CRTOD. When the CRDAY and CRTOD classes of service are enabled, CRDAY takes precedence.

When incoming calls require redirection, the order of precedence for which the system handles no answer and busy calls, is listed below:

Calls to Idle Stations:

Call Forward All Calls

Message Waiting

Call Forward No Answer

Attendant Recall

Calls to Busy Stations:

Call Forward All Calls

Hunting

Call Waiting or Camp-On

Message Waiting Forward Busy

Call Forward Busy

Call Forward, Internal

Call Forward/Hunt Override Via FFC

Call Forward No Answer

When Call Redirection by Day (CRDAY) is activated, calls unanswered after a specified number of rings are sent to Call Forward No Answer feature (CFNA) and forwarded to the alternate DN based on the alternative days/ holidays configured for the set.

Call Forward All Calls

All unanswered incoming calls are sent to the Call Forward All Calls feature if there is no answer then the call is sent to the alternate CRDAY DN specified for that day/holiday.

Call Waiting Redirection

When Call Forward No Answer (CFNA) occurs on a waiting call, the Call Waiting Redirection (CWTR) feature redirects the call to a specified DN based upon the alternate days and/or holidays configured for the set. If the current day matches one of the busy set's alternate days or holidays, then the call is redirected to the CRDAY alternate DN.

Hunting

Hunting allows a call encountering a busy DN to route automatically to another DN. When CRDAY is enabled and an incoming call reaches a busy DN, the current day is checked against the alternate days and holidays specified for that set. If the current day matches one of the alternate days, the call begins the hunting route using the alternate redirection DNs defined for the set.

Hunting by Call Type

The Hunt by Call Type redirects an incoming call to a HUNT DN. With CRDAY enabled on the called DN, the incoming calls on specified Alternate Days and Holidays are redirected to the assigned alternate HUNT DN.

Group Hunting

The Group Hunting feature provides a method of hunting DNs in a group. The group is associated with a Pilot DN, that is, a DN with no associated Terminal Number (TN). The hunting is done in the order of entry of DNs in the group. If a set's list of alternate days and/or holidays program matches the current day, incoming calls are directed to Group Hunt. This feature allows an incoming DID call to be redirected to a Hunt DN or External Hunt (EHT) if Call Forward by Call Type (CFCT) is enabled. To activate this feature, the called DN must have the following class of service Hunting Denied HTD with Hunting By Call Type Allowed (HBTA), and package 131 enabled.

Multiple Appearance DN, Multiple Appearance Redirection Prime

When a call redirection feature is activated for a Multiple Appearance DN (MADN), the TN information is required. Call redirection always refers to the Multiple Appearance Redirection Prime (MARP) TN to determine the feature operation. The CRDAY feature also uses the MARP TN to get the alternate call redirection DNs.

Second Level Call Forward No Answer

The Second Level Call Forward No Answer (SFNA) allows unanswered calls to receive Call Forward No Answer (CFNA) treatment twice. The CFNA timer is configured in the customer data block (CDB) for the number of rings before a call is redirected. If SFNA is allowed on the last DN rung, then the incoming call is redirected based on the list of alternate days and/or holidays configured for the set (day/holiday class of service).

User Selectable Call Redirection

The CRDAY feature does not support User Selectable Call Redirection (USCR). Only the following redirection DNs can be changed from a set:

- CFNA DN (FDN)
- External CFNA DN (EDN)

- Hunt DN (HUNT)
- External Hunt DN (EHT)

The alternate redirection DNs cannot be changed with USCR.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure alternate days/holidays.

2 LD 10 – Enable call redirection by day/holiday for analog sets.

3 LD 11 – Enable call redirection by day/holiday for a digital sets.

LD 15 – Configure alternate days/holidays.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Redirection data
CUST	0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
CRDAY	YES	Call Redirection by Day. DAY0 is prompted if “YES” is entered. (NO) = default.

- DAY0	x x...x	List of alternate days in list 0. Where x = 1...7 Sunday = 1 Monday = 2 Tuesday = 3 Wednesday = 4 Thursday = 5 Friday = 6 Saturday = 7 To remove a day value precede the day number with an X.
- DAY1	x x...x	List of alternate days in list one.
- DAY2	x x...x	List of alternate days in list two.
- DAY3	x x...x	List of alternate days in list three.
CRHOL		Call Redirection by Holiday.
	NEW CHG OUT OUT ALL	Add new data. Change existing data. Remove existing data. Delete all holidays in the list. There are a maximum of 20 holidays allowed. The four holiday options lists are created from these original 20 holidays.
- DATE	dd mm yyyy	Enter holiday date. dd = day. mm = month. yyyy = year (optional, with maximum year value of 2104). If the year is not entered, the holiday is repeated every year.
--HOL_OPT		Holiday Option List to which entered date applies.
	n n n n	Where: n = 0, 1, 2, or 3.
	ALL	Select all four holiday option lists. Precede the holiday list with X to remove.

- DATE	dd mm yyyy	Continue to input holidays to maximum of 20.
	<CR>	Stop adding holidays and continue administration.
--HOL_OPT	n n n n	Holiday Option List to which entered date applies. Where: n = 0, 1, 2, or 3. Select all four holiday option lists. Precede the holiday list with X to remove.

LD 10 – Enable call redirection by day/holiday for analog sets.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	 500	Set type. 500/2500 set data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		
CLS	FNA FBA RBDA RBHA	Call Forward No Answer Allowed. Call Forward Busy Allowed. Redirection By Day Allowed. Redirection By Holiday Allowed.
...		
ADAY	(0) - 3	Alternate days in DAY list 0-3 are selected for the set. Enter the list of alternate days listed in the Customer Data Block.
AHOL	(0) - 3	Enter the list of alternate redirection holidays in Holiday list 0-3 as selected for the set defined in the Customer Data Block.

...		
FTR	AFD x...x	Alternate forward DN to which all internal calls on an alternate day and/or holiday are redirected upon no answer.
	AHNT x...x	Alternate hunt DN to which all internal calls on an alternate day and/or holiday are redirected when set is busy or no answer. (CFNA to HUNT).
	AEFD x...x	Alternate external forward DN to which all external calls on an alternate day and/or holiday are redirected upon no answer.
	AEHT x...x	Alternate external hunt DN to which all external calls on an alternate day and/or holiday are redirected when set is busy or no answer (CFNA to HUNT).

LD 11 – Enable call redirection by day/holiday for a digital sets.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	 xxxx	Type of set: Where xxxx = 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		
CLS	FNA FBA RBDA RBHA	Call Forward No Answer Allowed. Call Forward Busy Allowed. Redirection By Day Allowed. Redirection By Holiday Allowed.
...		

ADAY	(0) - 3	Alternate days in DAY list 0 - 3 are selected for the DN.
AHOL	(0) - 3	Enter the list of alternate redirection holidays in Holiday list 0-3 as selected for sets defined in the Customer Data Block.
...		
AFD	x...x	Alternate forward DN to which all internal call forward DNs on alternate days/and all holidays are redirected upon no answer.
AHNT	x...x	When the DN is busy all incoming calls in the HOLIDAY list are redirected to the alternate hunt DN.
AEFD	x...x	Alternate external forward DN to which incoming calls are redirected on alternate days.
AEHT	x...x	When the DN is busy, all incoming calls for the specified holiday are redirected to the alternate external hunt DN.

Feature operation

No specific operating procedures are required to use this feature.

Call Redirection by Time of Day

Contents

This section contains information on the following topics:

Feature description 811
Operating parameters 812
Feature interactions 812
Feature packaging 814
Feature implementation 814
Feature operation 817

Feature description

Call Redirection by Time of Day (CRTOD) adds flexibility to the existing operations of Call Forward No Answer, Hunting and Call Forward by Call Type by allowing incoming calls to be automatically redirected to a predefined Directory Number at a specified time of day.

When the Call Redirection by Time of Day (CRTOD) feature is activated, incoming calls are automatically redirected to a Directory Number through Hunting, Flexible Call Forward No Answer, External Hunt or External Call Forward No Answer. Depending on the time of day, an incoming call can also be redirected to an alternate Directory Number using the Hunting, Call Forward No Answer and Call Forward by Call Type operations.

This feature only changes which redirection Directory Number or alternative Directory Number is used to redirect a call when possible.

Operating parameters

All existing limitations that apply to the operation and redirection of Directory Numbers of Call Forward No Answer and Hunting also apply to this feature.

Only one alternate time option is allowed per telephone set at a given time.

This feature is not supported on Basic Rate Interface (BRI) terminals.

Feature interactions

When incoming calls require redirection, the order of precedence for which the system handles no answer and busy calls, is listed below:

Calls to Idle Stations:

1 Call Forward All Calls

2 Message Waiting

3 Call Forward No Answer

4 Attendant Recall

Calls to Busy Stations:

1 Call Forward All Calls

2 Hunting

3 Call Waiting or Camp-On

4 Message Waiting Forward Busy

5 Call Forward Busy

Call Forward All Calls
Call Forward, Internal
Call Forward/Hunt Override Via FFC

These features take precedence over Call Redirection by Time of Day (CRTOD).

Call Forward by Call Type
Hunting by Call Type

If Call Forward by Call Type (CFCT) is enabled with Call Forward No Answer (CFNA) and Call Redirection by Time of Day (CRTOD), unanswered internal calls receiving CFNA are routed to the Flexible CFNA DN, Hunt DN, Alternate Flexible CFNA DN or Alternate Hunt DNs. External calls are routed in the same manner.

If CFNA is enabled with Hunting by Call Type and Call Redirection by Time of Day (CRTOD), unanswered internal calls are redirected to the Hunt DN or Alternate Hunt DN during the alternative time. External calls are routed in the same manner. The alternate time is defined on the called DN's data block.

Call Forward No Answer

Call redirection parameters for Call Forward No Answer are obtained from the originally dialed Directory Number. When CRTOD is activated, unanswered calls given CRTOD treatment are forwarded with CFNA according to the time of day. No changes are made to the existing CFNA feature.

Call Forward No Answer, Second Level

Existing Second Level CFNA allows unanswered calls to receive Call Forward No Answer treatment twice. CRTOD parameters are obtained from the last rung Directory Number. A maximum of two levels of CFNA is allowed for an unanswered call.

Call Waiting Redirection

When Call Forward No Answer occurs on a waiting call, the redirected Directory Number used depends on the time of day if CRTOD is activated.

Multiple Appearance DN Redirection Prime

When CRTOD and Multiple Appearance DN Redirection Prime (MARP) are activated, Call Forward or Hunt are dependent on the time of day and follows the MARP feature for Call Forward No Answer or Hunt treatment.

Hunting

When CRTOD is enabled and an incoming call reaches a busy Directory Number, the time is checked against the Alternate Redirection Time Option range defined on the telephone.

User Selectable Call Redirection

User Selectable Call Redirection is not supported.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 - Configure Alternative Redirection Time.
2. LD 10 - Configure Terminal Number Block for Analog (500/2500 type) telephones.
3. LD 11 - Configure Terminal Number Block for proprietary telephones

LD 15 - Configure Alternative Redirection Time.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Change Call Redirection.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

- CRTOD	YES	Call Redirection by Time of Day. Alternate time option prompts are skipped if (NO) or <CR> is entered
- - CRT0	SH SM EH EM	Alternate time option 0, where: SH = starttime in hours, SM = starttime in minutes, EH = endtime in hours and EM = endtime in minutes in international time format (hour from 00-23 and minute 00-59). Enter “X” to remove current value and reset both the start time and end time equal to 0.
- - CRT1	SH SM EH EM	Alternate time option 1, where: SH = starttime in hours, SM = starttime in minutes, EH = endtime in hours and EM = endtime in minutes in international time format (hour from 00-23 and minute 00-59). Enter “X” to remove current value and reset both the start time and end time equal to 0.
- - CRT2	SH SM EH EM	Alternate time option 2, where: SH = starttime in hours, SM = starttime in minutes, EH = endtime in hours and EM = endtime in minutes in international time format (hour from 00-23 and minute 00-59). Enter “X” to remove current value and reset both the start time and end time equal to 0.
- - CRT3	SH SM EH EM	Alternate time option 3, where: SH = starttime in hours, SM = starttime in minutes, EH = endtime in hours and EM = endtime in minutes in international time format (hour from 00-23 and minute 00-59). Enter “X” to remove current value and reset both the start time and end time equal to 0.

LD 10 - Configure Terminal Number Block for Analog (500/2500 type) telephones.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Analog telephone can be defined or modified.
TYPE:	500	Analog (500/2500 type) telephone data block.
...		

CLS	RTDA	Call Redirection by the Time of Day allowed. If CLS = RTDD (denied) then RTDA, AEFD, AEHT, AFDN, AHNT will be removed and ARTO prompt will be reset to 0.
...		
ARTO	(0)-3	Alternate Redirection Time Option for call redirection defined in the Customer Data Block. Only prompted if CLS = RTDA. Default value 0 is entered if request is new. The value is not changed if request is CHG. <CR> to enter CLS and ARTO data.
FTR	xxxx yyyy	Enter Feature Name and Related Data.
	AFD	Alternate Call Forward No Answer DN up to 13 digits. Remove by setting CLS = RTDD.
	AHNT	Alternate Hunt DN up to 13 digits. Remove by setting CLS = RTDD.
	AEFD	Alternate External Call Forward No Answer DN up to 13 digits. Remove by setting CLS = CFTD or RTDD.
	AEHT	Alternate External Hunt up to 13 digits. Remove by setting CLS = CFTD or RTDD.

LD 11 - Configure Terminal Number Block for proprietary telephones

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type, where xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS	RTDA	Call Redirection by the Time of Day allowed.
...		

ARTO	(0)-3	Alternate Redirection Time Option for call redirection defined in the Customer Data Block. Only prompted if CLS = RTDA. Default value 0 is entered if request is new. The value is not changed if the request = CHG. <CR> to enter CLS and ARTO data.
AFD	xxxx	Alternate Call Forward No Answer DN up to 13 digits. Remove by setting CLS = RTDD.
AHNT	xxxx	Alternate Hunt DN up to 13 digits. Remove by setting CLS = RTDD.
AEFD	xxxx	Alternate External Call Forward No Answer DN up to 13 digits. Remove by setting CLS = CFTD or RTDD. Requires Call Forward by Call Type Allowed (CFTA) Class of Service.
AEHT	xxxx	Alternate External Hunt up to 13 digits. Remove by setting CLS = CFTD or RTDD. Requires Call Forward by Call Type Allowed (CFTA) Class of Service.

Feature operation

No specific operating procedures are required to use this feature.

Call Transfer

Contents

This section contains information on the following topics:

Feature description 819
Operating parameters 820
Feature interactions 820
Feature packaging 830
Feature implementation 831
Feature operation 832

Feature description

The Three-Party Service Allowed Class of Service, part of the Multi-Party Operations feature, cannot be used together with the XFR Class of Service.

With the Restricted Call Transfer feature enabled, users of analog (500/2500 type) telephones cannot transfer calls. Attempted call transfers are not routed to the attendant.

Call Transfer allows a telephone user on any two-party call to hold the existing call and originate another call to a third party. The user may consult privately or transfer the original call to the third party. A call is transferred by pressing a dedicated key on Meridian 1 proprietary telephones or by flashing the switchhook on analog (500/2500 type) telephones.

Restricted Call Transfer

The Restricted Call Transfer feature provides the Call Transfer Restricted (XFR) Class of Service for analog (500/2500 type) telephones. By assigning XFR Class of Service in LD 10, a call transfer attempt will not result in action. This is different from the Call Transfer Denied (XFD) Class of Service, which will route the call to the attendant when a transfer is attempted.

Operating parameters

A separate Call Transfer key/lamp pair must be assigned to Meridian 1 proprietary telephones.

A transfer allowed Class of Service must be specified for analog (500/2500 type) telephones to access this feature.

If trunks are involved, successful completion of a transfer depends on the access restrictions assigned to the stations and trunks.

While the originating side of a call is linked to a transfer or conference key (that is, the originator of a transfer/conference call has not yet completed the transfer/conference), the terminating side cannot initiate a transfer or conference. Conference calls cannot be transferred.

Feature interactions

Advice of Charge for EuroISDN

When a set is connected to an ISDN CO trunk conveying AOC charging information, the received call charging information is stored against this set.

If the user transfers the call while the dialed set is still ringing, call charging information is stored against the transferring set until the call is either answered or abandoned by the external party. If the user consults with the dialed transfer set, charging information is stored against the transferring set until the call is either answered or abandoned. If the transferred call is redirected by a call redirection feature, the call is charged against the transferring set until the call transfer is completed and the call is answered. In all instances, if the call is answered, new call charging information is stored against the set receiving the transferred call.

AC15 Recall: Transfer from Norstar

A party involved in a consultation call (an active or held party) cannot initiate a consultation call for preventing call chaining. This principle is maintained in the following cases:

- the party is an AC15 trunk (if it attempts to initiate a consultation call, the recall signal is ignored), and
- the party is a local set, but the consultation call is made by an AC15 trunk.

Attendant Break-In

Until a transferred call is connected, the attendant cannot break in to a call that is being transferred.

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Call Transfer (TRN) key is ignored.

Call Forward by Call Type

Calls modified by Call Transfer receive Call Forward by Call Type treatment. If party A (telephone or trunk) calls party B, and B transfers to party C, the forwarding DN and Class of Service are obtained from party C

Call Forward, Break-In and Hunt Internal/External Network Wide

The treatment of a call following a call transfer (Call Forward/Hunt by Call Type) is based on the transferring set and the call originator's set. The set display on network call modification or redirection does not change.

Call Forward/Hunt Override Via Flexible Feature Code

A set can activate Call Forward/Hunt Override Via FFC when initiating a transfer. If the transfer is completed while ringing, the Call Forward/Hunt Override will still be active and passed on to the transferred party.

Call Hold, Deluxe

A consultation call can be placed on Hold.

Call Page Network Wide

A station set or Attendant Console that transfers an external Call Page Network Wide (PAGENET) uncontrolled call is not blocked. However, an external PAGENET controlled call is blocked.

Call Party Name Display

When the Transfer key is pressed during an active call, the display clears. The call is in a held state. The DN and name of the transferred telephone appear on the display when the DN is dialed. When the transfer is complete, the transferring telephone's display clears because the telephone is now disconnected. The transferred telephone's display changes to show the name of the newly connected party.

Call Pickup Network Wide

A call may be picked up before or after the transferring party has completed the transfer.

For pickup before transfer completion, the transferring party is displayed updated information by the Call Pickup Network Wide feature when the call is picked up. Then, when the transfer is completed, normal call transfer information is exchanged by each party involved in the final call.

For pickup after call transfer completion, everything happens as if the call had been made directly from calling to ringing party. After pickup is performed, displays are updated as for normal Call Pickup.

Call Transfer

Call Completion notification is only presented to the Call Completion originating set. This notification cannot be transferred to another station. Once the second call is completed, the call can be transferred.

If a user encounters a busy or no answer situation during a transfer operation, Call Completion can be activated.

Calling Party Name Display Denied

During a Call Forward or Call Transfer, the calling party digits and forwarding/transferring party digits are displayed on the terminating set. This display is allowed or denied depending on the Class of Service of the calling set and the forwarding/transferring set. The name of the forwarding/ transferring set is not displayed on the calling and terminating set.

Calling Party Privacy

If an incoming non-ISDN call is being transferred or an incoming ISDN call is transferred to a non-ISDN trunk, the Calling Party Name and Number will not be passed on to the terminating set. The CPP feature will not change this operation.

For cases where an incoming call with the Privacy Indicator is transferred over an MCDN trunk, or to a local station, the name and/or number of the originating party will not be displayed on the set of the final terminating party.

Charge Account and Calling Party Number

A Call Transfer call produces two records: a Call Detail Recording(CDR) start record and a CDR end record.

Charge Account, Forced

If an FCA code is entered at the beginning of a call, the new unrestricted Class of Service remains in effect for any transfer or conference made during the call. If all FCA criteria are met, an account number entered after activating the Conference key, Call Transfer key, or a switchhook flash is interpreted as an FCA code.

China – Attendant Monitor

If any party at the customer location involved in a monitored call attempts to activate call transfer, monitoring is immediately deactivated.

China – Toll Call Loss Plan

Toll pad switching is also provided after call transfer has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Conference

Conference can be used to transfer calls, eliminating the need for a separate Call Transfer key/lamp pair on Meridian 1 proprietary telephones. Calls in the ringing state cannot be transferred with Conference. The third party must answer before the transfer can be completed.

A conference can also be established after initiating a Call Transfer operation. After the third party answers, pressing the Conference key establishes a three-way conference.

When a switchhook flash transfers calls on analog (500/2500 type) telephones with three-party conference (C6A) Class of Service, the transferring party goes on hook, leaving the other two parties established. Telephones with a C6A Class of Service involved in a conference having more than three parties must add the last party to the conference, then flash the switchhook and go on hook to complete the transfer.

Dial Intercom

When using Conference or Transfer, the voice option is not provided if the call is terminated before the conference or transfer is completed. If an analog (500/2500 type) telephone is part of a Dial Intercom Group (DIG), the user of the telephone can conference only with another user whose telephone is within the same Dial Intercom Group (DIG).

Display Calling Party Denied

When a set transfers a call, display information is updated according to the Class of Service of the respective sets. This occurs for both internal and ISDN network calls.

If an unsupervised call transfer occurs on an internal call, the DN of the terminating set is displayed to the calling party regardless of the DPD Class of Service options that are configured on the terminating set.

Dial Access to Group Calls

Group Call

Call Transfer cannot be applied to Dial Access to Group Calls or Group Call.

Group Hunt

Any call may be transferred to a Group Hunt Pilot DN. If there are no idle sets available for the call transfer, the call is queued to the Pilot DN and the caller receives ring back tone. If the call cannot be queued because the queue threshold has been reached, the caller receives busy tone.

Group Hunting Queuing Limitation Enhancement

If a call is transferred to the PLDN, and all Group Hunt list members are busy, the call is queued to the PLDN, if the number of queued calls is less than the Group Hunt Threshold limit. If the number of queued calls has reached the Group Hunt Threshold limit, the call is not queued and busy tone is returned to the transferring party.

Held Call Clearing

Active Call Transfer calls are cleared by either an on-hook or Release key action. Held Call Transfer calls are cleared only by an on-hook action, and not by a Release key action.

Hold

A consultation call can be placed on Hold.

In-Band Automatic Number Identification

If an agent transfers an In-Band Automatic Number Identification (IANI) call to another Automatic Call Distribution DN, the ANI number is displayed on the terminating set's display.

ISDN QSIG/EuroISDN Call Completion

Call Completion notification is only presented to the Call Completion originating set. This notification cannot be transferred to another station. Once the second call is completed, the call can be transferred.

If a user encounters a busy or no answer situation during a transfer operation, Call Completion can be activated.

ISDN QSIG Name Display

After the completion of a call transfer, an incoming QSIG call with name display presentation allowed has name information displayed on the destination set. If the incoming QSIG call has presentation denied, name information is not displayed to the destination set.

Loop Start Supervisory Trunks

If an internal station user transfers an answered outgoing call to another station in the ringing state, then any disconnect signal received from the far end causes the trunk to be released and ringing of the internal set to stop. This operation eliminates the problem of holding trunks and extensions due to lack of supervision on Loop Start trunks.

Meridian 911

Trunk priority associated with an incoming 911 call is only preserved if blind transfer is used.

Meridian 911 - Call Abandon

M911 abandoned calls cannot be transferred.

Message Registration

The party that originates a call is charged. The charge cannot be moved to another party using Call Transfer.

Multi-Party Operations

Analog (500/2500 type) telephones with TSA Class of Service can Call Transfer by going on-hook after establishing a conference. This differs from operation with XFA Class of service, where transfer can be achieved by going on-hook during Consultation connection.

If an analog (500/2500 type) telephone with TSA Class of Service goes on-hook during consultation connection, it is treated as misoperation of All Other Cases and the recovery actions are done based on the CCDO and AOCS options selected in LD 15. If CDOC = NO, an analog (500/2500 type) telephone can achieve a transfer by going on-hook after establishing a conference.

During the Consultation connection, the non-controlling parties are restricted from using Call Transfer, Conference and Three-party Service features.

Multi-Party Operations Enhancements

A party receiving Patience Tone or recall of misoperation ringback is not able to Call Transfer.

Music, Enhanced

The held party receives Music when the other party presses the Call Transfer key. The Music connection remains until the Call Transfer key or the DN key is pressed, ending the Consultation Hold state.

Network Intercom

Hot Type I calls may be transferred to another Hot Line key or to a normal DN key; likewise calls on a normal DN key may be transferred to a Hot Line key.

Off-Hook Alarm Security

A telephone receives the Off-Hook Alarm Security treatment if the telephone has ASCA Class of Service and attempts to transfer a call and the ASTM expires.

On Hold on Loudspeaker

It will not be possible to transfer the loudspeaker call to another party.

Periodic Pulse Metering

If the user of a station which is connected to a metered trunk transfers an internal call to another internal station while the dialed station is still ringing, the Periodic Pulse Metering (PPM) pulse count is accumulated against the transferring station until the call is answered by the dialed party, or abandoned by the dialing party. When the call is answered, the pulses are counted against the station to which the call has been transferred. If the station user transfers the call after consulting with the dialed station user, then the PPM pulses are counted against the controlling station until the call is transferred. When the call is transferred, the PPM pulses are counted against the station to which the call has been transferred. If the transferred call is redirected using any of the call redirection features such as Call Forward or Hunting, the call is charged against the transferring station until the call is transferred. The pulses are then counted against the answering station. This method ensures that PPM meters are charged in a manner consistent with the printing of CDR records.

Predictive Dialing

The application sends the Fast Transfer request on behalf of a Meridian 1 proprietary telephone, and then the switch initiates and completes the transfer immediately which is similar to a normal call transfer from a Meridian 1 proprietary telephone.

In a Predictive Dialing scenario where the autodialer (origination DN) is a Meridian 1 proprietary telephone, the Make Call message sent by the application to the switch to make a call on behalf of the Meridian 1 proprietary telephone, and then the call transfer call, will interact with the Meridian 1 proprietary telephone Call Transfer feature. The autodialer is configured with Class of Service TRN so that the switch can transfer the call to the target destination.

The application sends the Fast Transfer request on behalf of an analog (500/2500 type) telephone. The switch will then initiate and complete the transfer in one step.

In a predictive dialing scenario, the application will send the Make Call request on behalf of the autodialer (analog (500/2500 type) telephone) to have the switch make the call, and then transfer the call when the switch receives the Fast Transfer message. The autodialer needs to be configured with Classes of Service Dial Pulse (DIP) and Transfer Allowed (XFA) for 500 sets, or with Classes of Service Digitone (DTN) and XFA for 2500 sets.

Privacy Override

Calls in a Privacy Override conference state cannot be transferred.

Station Activity Records

A Station Activity Record is generated when a set with Class of Service Call Detail Monitoring Allowed (CDMA) transfers a trunk call. CDR “X” record generation is not affected by this development. The set to which the call is transferred also produces a Station Activity Record if it has Class of Service CDMA and answers the call. When the second “D” record is produced (by the set to which the call is transferred), the digits field of the “D” record shows the digits dialed by the transferring set.

Supervised Analog Lines

China-Supervised Analog Lines

If more than one active call is extended to an analog line, the call type associated with an analog line is determined by the first active call. The call type is assumed to be incoming and hook flash supervision applies if a terminal device answers an incoming call from an idle state. If the terminal device performs a switch hook flash to put the first party on hold and initiates a consultation call, the Battery Reversal feature is not supported; no battery reversal answer signal is extended to the terminal device when the second party answers.

If the first party disconnects while the terminal device is connected to the second party, no disconnect supervision is extended to the terminal device. However, hook flash disconnect supervision is extended to the terminal device when the second party disconnects (that is, a disconnect supervision signal is sent only when the last party connected to the terminal disconnects).

If a terminal device originates an outgoing call, battery reversal answer supervision is extended when the called party answers. The polarity of the line remains reversed polarity when the terminal device performs a switch hook flash and then initiates a consultation call to a second party. The analog line is reverted to normal polarity when the terminal device completes the transfer and drops out or when the last of either the held party or the consultation party disconnects.

Three-Party Service

The party receiving the patience tone or the Misoperation ringback is not able to make a call transfer.

Trunk Barring

The originator of a call transfer, unless otherwise restricted, is able to connect to a denied party on a consultation basis. Operating the Transfer key on a Meridian 1 proprietary telephone or going on hook on an analog (500/2500 type) telephone does not result in a call transfer if the Originating Trunk Connection is barred. The user of a Meridian 1 proprietary telephone remains connected to the denied party until releasing the connection and returning to the held Originating Trunk Connection. The user of an analog (500/2500 type) telephone is re-rung by the Originating Trunk Connection when transfer is attempted and denied.

Trunk to Trunk Connection

To transfer an external trunk on ringing across a supervised analog network TIE trunk, the external trunk and internal TIE line must have both answer and disconnect supervision, and the external call must be established. To transfer one outgoing external trunk to another, both external trunks must have answer and disconnect supervision, and both external calls must be established.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Configure Call Transfer for analog (500/2500 type) telephones.

2 LD 11 – Add a Call Transfer key for Meridian 1 proprietary telephones.

3 LD 10 – Restricted Call Transfer for an analog (500/2500 type) telephone.

LD 10 – Configure Call Transfer for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA	(Deny) allow Call Transfer.

LD 11 – Add a Call Transfer key for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx TRN	Add a Call Transfer key (the M2317 and M3000 must use key 26).

LD 10 – Restricted Call Transfer for an analog (500/2500 type) telephone.

Prompt	Response	Description
TYPE:	NEW CHG	Add new data. Change existing data.
REQ:	500	Telephone type.
...		
CLS	XFR	Restrict call transfers and do not recall to attendant.

Feature operation

To transfer an active call on a Meridian 1 proprietary telephone, follow these steps:

1. Press **Transfer**.
   The call is on hold.
2. Dial the number where you want to transfer the call.
3. Press **Transfer** when you hear ringing or after your call is answered.
   When your call is answered, you may speak privately with the new party before completing the transfer.

   Note: To cancel an incomplete transfer, press the key beside the fast flashing indicator and you return to the call you tried to transfer. To conference all three parties, press the Conference key, if equipped.

To transfer an active call on an analog (500/2500 type) telephone, follow these steps:

1. Flash the switchhook.
   The call is on hold.
2. Dial the number where you want to transfer the call.
3. Flash the switchhook when you hear ringing or after your call is answered.

   When your call is answered, you may speak privately with the new party before completing the transfer.

Note: To cancel an incomplete transfer, hang up, then lift the handset and flash the switchhook to return to the call.

Call Waiting Redirection

Contents

This section contains information on the following topics:

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Operating parameters 836
Feature interactions 836
Feature packaging 842
Feature implementation 842
Feature operation 845

Feature description

Previously, Call Waiting notified an active set that a second call was waiting to be answered on that Directory Number (DN). For non-attendant extended calls, the incoming call received Call Waiting treatment until the call was answered by the called party or the calling party disconnected. For attendant-extended calls, the incoming call received Call Waiting treatment until the Call Waiting Recall timer timed out, at which time the call was recalled to the attendant. The attendant had to then extend the call to a message center or voice mail. However, since the attendant was given no indication as the reason for the recall (called party busy or not answering), it was difficult for the attendant to redirect the call properly.

The Call Waiting Redirection feature follows the Call Forward No Answer (CFNA) treatment defined for the DN. No modifications have been made; all existing Call Forward No Answer functionalities will apply to redirected calls.

Operating parameters

The existing Call Waiting and Call Forward No Answer limitations apply to the Call Waiting Redirection feature. The Call Forward No Answer feature is used by the Call Waiting Redirection feature to redirect “no answer” calls given Call Waiting treatment.

Although the Call Waiting treatment is applied to a busy DN, the CFNA call redirection treatment given by the Call Waiting Redirection feature is for a “no answer” presentation. The unanswered Call Waiting call is treated as a call presented to an idle “no answered” DN. Calls redirected to messaging services or sets with displays are provided with the “no answer” call redirection reason.

The existing implementation of Call Forward No Answer is used to select the TN with the CFNA DN for the “no answer” Call Waiting call. Calls are redirected according to the call type (internal or external) as defined at the designated call redirection TN chosen by CFNA.

Feature interactions

Automatic Timed Reminders
Automatic Timed Recalls

When CFNA is active, the Slow Answer Recall timer begins only after the call reaches its final destination. CFNA has precedence over Attendant Recall for attendant-extended calls. Irrespective of the relative time-out intervals for each feature, ringing continues as long as allowed by CFNA for sets with CFNA enabled.

Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also has precedence over the Call Waiting recall timer.

Basic Rate Interface

The Call Waiting Redirection feature is not applicable to Basic Rate Interface (BRI) terminals. However, an ISDN BRI terminal may redirect a call using hunting or CFNA.

Call Forward All Calls
Call Forward, Internal

Call Forward All Calls and Internal Call Forward both have precedence over Call Waiting and the Call Waiting Redirection feature.

Call Forward and Hunt by Call Type

If Call Forward and Hunt by Call Type (CFCT) is enabled with Call Forward No Answer and Call Waiting Redirection, “no answer” internal calls receiving Call Waiting treatment are routed for CFNA treatment to the Flexible CFNA DN (FDN) or Hunt DN, and “no answer” external calls are routed for CFNA treatment to the External Flexible CFNA DN (EFD) or External Hunt DN (EHT).

Call Forward/Hunt Override via Flexible Feature Code

There is no interaction with the Call Waiting treatment component of the Call Waiting Redirection feature. However, Call Forward/Hunt Override via Flexible Feature Code does override CFNA, and thus the CFNA treatment given to unanswered Call Waiting calls by the Call Waiting Redirection feature is overridden by the Call Forward/Hunt Override via Flexible Feature Code (CFHO) feature. The incoming call will continue to be given Call Waiting treatment as if the Call Waiting Redirection feature is disabled when the CFHO feature is enabled by the calling party.

Call Forward No Answer

Per existing Call Forward No Answer (CFNA) feature operation, the call redirection parameters for CFNA are obtained from the originally dialed DN for redirected calls.

Existing Network CFNA treatment is applied to calls receiving Call Waiting treatment on sets with CFNA and the Call Waiting Redirection feature enabled if the Call Waiting call is not answered before the expiration of the CFNA timer and the CFNA DN is on another node.

Call Forward No Answer, Second Level

The existing Second Level Call Forward No Answer treatment is applicable to Call Waiting calls redirected by CFNA (first level) with the Call Waiting Redirection feature which are still not answered at the last rung DN.

Call Redirection by Time of Day

When Call Forward No Answer occurs on a waiting call, the redirected Directory Number used depends on the time of day if Call Redirection by Time of Day (CRTOD) is activated.

Call Pickup Network Wide

A call that is redirected by the Call Waiting Redirection feature to the active set's Call Forward No Answer DN may be picked up.

Call Waiting

The option selected for Call Waiting Redirection treatment also applies to calls given Internal Call Waiting treatment.

Camp-On

There is no interaction as Call Waiting and Camp-On are mutually exclusive.

Direct Inward Dialing Call Forward No Answer Timer

The Direct Inward Dialing Call Forward No Answer Timer timer is applied after the last stage of CFNA or SFNA treatment resulting from the Call Waiting Redirection feature for DID Call Waiting calls.

Distinctive Ringing/New Distinctive Ringing Call Forward No Answer

The existing Distinctive Ringing Call Forward No Answer feature is applied to calls from a Distinctive Ringing enabled trunk. If such an incoming call is receiving Call Waiting treatment on sets with Distinctive Ringing, Call Forward No Answer (CFNA), and the Call Waiting Redirection feature enabled, the DFNA timer is applied to the call instead of the CFNA timer. The Call Waiting warning tone, if enabled, is not changed by Distinctive Ringing. If that call is not answered before the expiration of the DFNA timer, CFNA treatment is given using the Call Waiting Redirection feature.

Forward No Answer Call Waiting Direct Inward Dialing

With the Call Waiting Redirection feature also enabled, the Call Waiting Redirection feature takes precedence over Forward No Answer Call Waiting Direct Inward Dialing (FCWD). The existing CFNA also takes precedence over the existing Attendant Recall of Call Waiting calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call while the FCWD feature applies an attendant recall timer, the Call Waiting Redirection feature also has precedence over the FCWD timer.

Hunting

If Call Forward and Hunt by Call Type (CFCT) is enabled with Call Forward No Answer and Call Waiting Redirection, “no answer” internal calls receiving Call Waiting treatment are routed for CFNA treatment to the Flexible CFNA DN (FDN) or Hunt DN, and “no answer” external calls are routed for CFNA treatment to the External Flexible CFNA DN (EFD) or External Hunt DN (EHT).

Internal or Station-to-Station Call Waiting

The option selected for Call Waiting Redirection treatment also applies to calls given Internal Call Waiting treatment.

Meridian Mail Voice Mailbox Administration

Unanswered calls given Call Waiting treatment may now be allowed to forward to Voice Mail through the activation of the Call Waiting Redirection feature. The greeting given to the caller is for a “no answer” condition.

Message Center

Unanswered calls given Call Waiting treatment may now be allowed by the Call Waiting Redirection feature to be forwarded to a CFNA DN which may be a Message Center. The call redirection reason is “no answer”.

Message Waiting

Message Waiting has precedence over CFNA and Attendant Recall for attendant-extended calls. Unanswered calls given Call Waiting treatment are forwarded to the Flexible CFNA DN by the Call Waiting Redirection feature.

Multi-Party Operations

Recovery on Misoperation of Call Transfer – Call Transfer with Ring No Answer (RGNA)

If the transferring party goes on-hook to complete the Call Transfer (that is, blind transfer) before the "transferred to" or called party answers the Call Waiting call, an RGNA Misoperation of Call Transfer is detected by Multi-Party Operations (MPO).

With the Call Waiting Redirection feature enabled, if the blind transfer completes after the CFNA timer applied by Call Waiting Redirection has expired, there is no interaction as the Call Waiting Redirection feature is done and has already redirected the unanswered Call Waiting call to the CFNA DN.

If the blind transfer completes before the Call Waiting Redirection CFNA timer expires and the RGNA option is defined to be:

1 Standard (STD, that is, operation as it was prior to introduction of MPO), there is no interaction, as Call Waiting Redirection will redirect the unanswered Call Waiting call when the CFNA timer expires.

2 Non-STD (that is, ATN, DIS, OVF, AAR, or DAR), the RGNA option has precedence over CFNA and thus has priority over Call Waiting Redirection's CFNA treatment.

Recovery on Misoperation of Call Transfer – Misoperation of Call Transfer for All Other Cases

This type of misoperation occurs when the transferring party attempts to complete the transfer in several other non-RGNA scenarios. There is no interaction with these Multi-Party Operations scenarios and the Call Waiting Redirection feature.

Multiple Appearance Directory Number

The Call Waiting Redirection feature applies to unanswered Call Waiting calls which apply to single appearance DNs and primary appearance DNs of Multiple Appearance.

Multiple Appearance Directory Number Redirection Prime

If the Multiple Appearance Directory Number Redirection Prime (MARP) feature is activated, the Call Forward No Answer (CFNA) treatment given by Call Waiting Redirection for an unanswered Call Waiting call follows the MARP feature for CFNA treatment of calls to an idle DN.

Network Call Forward No Answer

Existing Network CFNA treatment is applied to calls receiving Call Waiting treatment on sets with CFNA and the Call Waiting Redirection feature enabled if the Call Waiting call is not answered before the expiration of the CFNA timer and the CFNA DN is on another node.

Network Call Redirection

Incoming calls receiving Call Waiting Redirection treatment can be redirected over the network only if the trunk-to-trunk connection is already supported per the existing Network Call Redirection feature.

Night Restriction Classes of Service

The Call Waiting Redirection feature applies to unanswered calls given Call Waiting treatment when the Night Restriction Classes of Service feature allows Call Waiting.

Night Service
Night Service Enhancements

Night Service has the same interaction with the Call Waiting Redirection feature as attendant-extended calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also has precedence over the Call Waiting recall timer.

Voice Mail

Unanswered calls given Call Waiting treatment may now be allowed to forward to Voice Mail through the activation of the Call Waiting Redirection feature. The greeting given to the caller is for a “no answer” condition.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the CFNA treatment, the number of ringing cycles for CFNA, and the Call Waiting Redirection option.

2 LD 10 – Configure Call Waiting and Call Forward No Answer for analog (500/2500 type) telephones.

3 LD 11 – Configure Call Waiting, and Call Forward No Answer for Meridian 1 proprietary telephones.

LD 15 – Configure the CFNA treatment, the number of ringing cycles for CFNA, and the Call Waiting Redirection option.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Call Redirection
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	CWRA	Call Waiting Redirection Allowed. Allow Call Forward No Answer treatment for unanswered Call Waiting calls on a DN.
...		
FNAD	(HNT) ATT NO FDN	CFNA treatment for DID calls.

FNAT	(HNT) ATT NO FDN	CFNA treatment for internal non-Direct Inward Dialing calls.
FNAL	(HNT) ATT NO FDN	CFNA treatment for calls when Call Waiting Redirection is enabled.
...		
CFN0 CFN1 CFN2	1-(4)-15 1-(4)-15 1-(4)-15	CFNA timers; number of normal ringing cycles for CFNA Options 0, 1, and 2.
DFN0 DFN1 DFN2	1-(4)-15 1-(4)-15 1-(4)-15	Distinctive Ringing CFNA timers; number of distinctive ringing cycles for CFNA Options 0, 1, and 2.

LD 10 – Configure Call Waiting and Call Forward No Answer for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
HUNT	xxx..x	Hunt DN. If the Call Forward No Answer treatments set up in LD 15 are set to HNT, the Hunt DN should be configured.
CLS	(CWD) CWA	Call Waiting external (denied) allowed.
	(SWD) SWA	Internal (Station-to-Station) Call Waiting (denied) allowed. If SWA is defined, CWA must also be defined.
	(WTA) WTD	Warning tone (allowed) denied.
	(FND) FNA	Call Forward No Answer (denied) allowed.

	FBA	Call Forward Busy Allowed. For customers with the United Kingdom (UK) package 190 configured, CLS must be set to FBA for calls over DASS/ DPNSS trunks.
RCO	(0)-2	Ringing cycle option for Call Forward No Answer. Prompted when CLS = FNA, or MWA (or both). RCO must be set to a value other than 0 for Call Waiting Redirection to operate.
FTR	aaaa FDN xxx ..x	Enter the feature and related data. Flexible CFNA DN. If the CFNA treatments set up in LD 15 are set to FDN, the Flexible CFNA DN should be configured.

LD 11 – Configure Call Waiting, and Call Forward No Answer for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, and 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FDN	xxx..x	Flexible CFNA DN. If the Call Forward No Answer treatments set up in LD 15 are set to FDN, the Flexible CFNA DN should be configured.
CLS	(SWD) SWA	Internal (Station-to-Station) Call Waiting (denied) allowed. SWA does not need to exist to have external Call Waiting.
	(WTA) WTD	Warning tone (allowed) denied.
	(FND) FNA	Call Forward No Answer (denied) allowed.
	FBA	Call Forward Busy Allowed. For customers with the United Kingdom (UK) package 190 configured, CLS must be set to FBA for calls over DASS/ DPNSS trunks.

RCO	(0)-2	Ringing cycle option for Call Forward No Answer. Prompted when CLS = FNA, or MWA (or both). RCO must be set to a value other than 0 for Call Waiting Redirection to operate.
HUNT	xxx..x	Hunt DN. If the Call Forward No Answer treatments set up in LD 15 are set to HNT, the Hunt DN should be configured.
KEY	xx CWT	xx = key number. Add a Call Waiting key (the M3000 must use key 24).

Feature operation

No specific operating procedures are required to use this feature.

Call Waiting/Internal Call Waiting

Contents

This section contains information on the following topics:

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Operating parameters 848
Feature interactions 849
Feature packaging 856
Feature implementation 856
Feature operation 857

Feature description

Call Waiting notifies a telephone user on an established call (internal or external) that an external call is waiting to be answered. Meridian 1 proprietary telephones must have a Call Waiting key/lamp pair assigned and a Class of Service that allows a warning tone. Call Waiting is applicable to the Prime DN or any single appearance DN on the telephone. When an external call comes into a Meridian 1 proprietary telephone and the telephone user is on a call, the Call Waiting lamp flashes and a buzz sounds through the telephone's speaker.

To use Call Waiting, analog (500/2500 type) telephones must have a Class of Service that allows Call Waiting and a warning tone. Two tone bursts are received through the handset to advise an analog (500/2500 type) telephone user of a waiting call. Note that the two calls cannot be conferenced together.

Call Waiting applies to Direct Inward Dialing (DID), Central Office (CO), Foreign Exchange (FX), and Wide Area Telephone Service (WATS) trunk calls extended to a busy telephone by the attendant. Call Waiting also applies to calls on TIE and Common Control Switching Arrangement (CCSA) trunks.

Internal Call Waiting

This feature provides Call Waiting for internal calls. This option, defined on a per-telephone basis, allows Call Waiting for calls from other telephones within the customer group. These calls include the following:

- direct telephone-to-telephone calls
- attendant-extended internal calls
- telephone-to-telephone call transfer of all trunk and internal calls

Call Waiting Flexible Feature Codes

A user may activate Call Waiting from an analog (500/2500 type) telephone with Call Waiting Class of Service by dialing the Call Waiting Activate (CWGA) FFC (defined in LD 57). To deactivate Call Waiting, the user dials either the Call Waiting Deactivate (CWGD) FFC (defined in LD 57) or the general Deactivate (DEAF) FFC (also defined in LD 57).

If Call Waiting is deactivated using FFCs, then station-to-station call waiting is also deactivated at the telephone.

If the Class of Service is CWA, ACTIV or DEACT will be printed in brackets when CWT is activated or deactivated.

The CWT FFCs do not affect Precedence Call Waiting.

Operating parameters

A Meridian 1 proprietary telephone can have only one working Call Waiting key/lamp pair.

Telephones with internal telephone-to-telephone Call Waiting must also have external Call Waiting (CWA) Class of Service.

A Call Waiting indication is not presented to a single-line telephone in the transfer or conference mode.

An analog (500/2500 type) telephone user receiving a second call can connect alternately to the original call and the Call Waiting call by a switchhook flash. The user cannot transfer or conference either call.

When a second call goes to a telephone that already has a Call Waiting call, the second call is recalled to the attendant if it is not answered by the number of rings defined in the Customer Data Block (RTIM prompt, field zz).

An analog (500/2500 type) telephone user who has received a Call Waiting call routed from the attendant cannot reconnect to the original call until it has been released from the console.

Attendant Administration does not support the Internal Call Waiting feature.

If a call is waiting and Call Waiting is deactivated using the Call Waiting Deactivate (CWGD) FFC, the call that is waiting is allowed to continue waiting while any new calls will not be allowed to wait.

Feature interactions

Advice of Charge for EuroISDN

When an Advice of Charge call is transferred to a busy set with Call Waiting Allowed, the transferring station is charged until the call is answered.

Attendant Blocking of Directory Number

If a set that has the Station-to station Call Waiting feature active (CLS SWA and a Call Waiting (CWT) key for SL-1 and digital sets) is idle when an Attendant Blocking of DN attempt is made, the Attendant Blocking of DN attempt will be allowed and processed as normal. If the DN is idle and there is an active call on the Call Waiting key, the Attendant Blocking of DN attempt will be allowed.

If a set has the Station-to-station Call Waiting feature active and the DN to be blocked is busy when an Attendant Blocking of DN attempt is made, the Attendant Blocking of DN attempt will be canceled and busy tone will be returned.

For a set that has the Call Waiting (or Station-to-station Call Waiting) feature active and a DN is blocked due to the Attendant Blocking of DN feature, any incoming call to the blocked DN will receive busy tone.

Attendant Break-In

If the destination DN has a camped-on incoming trunk call, the attendant cannot extend the urgent incoming call as a Camp-On call.

Attendant Incoming Call Indicators

The ICI feature is used with the Call Waiting feature to recognize, answer, and process incoming calls.

Attendant Queuing

Call Waiting options do not apply to calls queued to a specified attendant. The exception to this is the display call waiting key, which shows the number of calls in the overall attendant queue and the calls in the queue for a specified attendant.

Automatic Line Selection

A call on the Call Waiting key is not selected.

Call Forward All Calls
Call Forward All Calls takes precedence over Call Waiting.

Call Forward Busy - proprietary telephones

If Class of Service allows Call Forward Busy and Call Waiting Allowed, and the proprietary telephone has a Call Waiting key, calls do not forward to the attendant when the telephone is busy and another call is waiting.

Call Forward Busy - Analog (500/2500 type) telephones

If a telephone has Call Forward Busy and Call Waiting Allowed Class of Service, calls are forwarded to the attendant when the telephone is busy and has another call waiting.

Call Forward/Hunt Override Via Flexible Feature Code

Call Waiting can be used even if the Call Forward/Hunt Override Via FFC feature has been activated. When a busy set with Call Waiting configured is encountered, it will terminate on Call Waiting.

Call Forward, Internal Calls

Internal Call Forward takes precedence over Call Waiting.

Call Forward No Answer

If a call to a telephone gets Call Forward No Answer treatment to another telephone which is busy, Call Waiting and Camp-On do not apply. The call will attempt to terminate on the original DN again.

Call Park

A recall of a parked call is not presented in the Call Waiting mode. If an internal telephone is in the parked state, Call Waiting to that telephone is not provided.

Call Pickup Network Wide

Call Waiting calls cannot be picked up.

Call Waiting Redirection

The option selected for Call Waiting Redirection treatment also applies to calls given Internal Call Waiting treatment.

Camp-On

Call Waiting and Camp-On are mutually exclusive. If a Call Waiting Class of Service or key is defined, Camp-On cannot be provided.

Camp-on, Station

Call Waiting takes precedence over Station Camp-On.

China Number 1 Signaling - Called Party Control

An attendant cannot apply Call Waiting on an outgoing call that follows Called Party Control.

Dial Intercom

Call Waiting does not apply to a Dial Intercom appearance.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is permitted (consult-only state) into a requested party having call waiting.

Directory Number Delayed Ringing

Call Waiting tones apply to SCN/MCN keys as per existing operation. The Directory Number Delayed Ringing does not apply, and the user is informed of the incoming call immediately.

Flexible Feature Code Boss Secretarial Filtering

Call Waiting to a boss set with filtering active is routed to the secretary set.

Flexible Feature Code enhancement

The Call Waiting Deactivate (CWGD) FFC may be used to deactivate Call Waiting. If a call is waiting when Call Waiting is deactivated, the call is allowed to continue waiting while any new calls will not be allowed to wait.

Flexible Voice/Data Terminal Number

Call Waiting is not supported on data calls to a dynamic voice/ data TN.Call Waiting is supported for voice calls to dynamic voice/ data TN.

Group Hunt

Call Waiting to a Pilot DN will not be supported.

Hunting

If a call comes into a busy DN, it begins the hunting route defined from the called DN. If there are idle DNs on the hunting route, the call becomes a Call Waiting call on the called DN.

Idle Extension Notification

This feature can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When a busy set is encountered, it is possible to place an IEN request against the set.

ISDN Night Service

If a call is diverted to a third-party operator Night DN that is busy, Call Waiting may be activated (if equipped). The call to the third-party operator circuit switched network is released.

ISDN QSIG/EuroISDN Call Completion

On an Analog (500/2500 type) set, Call Completion notification waits until the set has finished an active call. If Call Waiting is configured on a set, notification is presented after the Call Waiting call. If an additional call is queued while Ring Again free notification is waiting on a set, the waiting call takes precedence over the Call Completion notification. An established Call Completion call is also queued if a set has Call Waiting feature equipped and is occupied on another call.

Message Center

Call Waiting calls are not forwarded to a Message Center.

Multi-Party Operations

A user of an analog (500/2500 type) telephone can answer a Call Waiting call, thereby establishing a consultation connection. The user can dial control digits, as normal. To toggle the calls, the toggle control digit must be used rather than a switchhook flash.

Multi-Party Operations

An analog (500/2500 type) telephone may be assigned both CWA and TSA Classes of Service. The user can establish a Consultation connection by answering Call Waiting during an active established call. If this is done, Control Digit features (CNFD, TGLD, and DISD) are available. Note that Programmable Control Digit TGLD, rather than a switchhook flash, is used to toggle the calls. Operation with XFA Class of Service is unchanged.

The Three-party Service feature changes the operation of Call Waiting for all analog (500/2500 type) telephones as follows (regardless of whether the sets have TSA Class of Service. If an analog (500/2500 type) telephone user activates Waiting during an active call so as to establish a Consultation connection, and if the user goes on-hook during the Consultation connection, the operation is treated as an AOCS misoperation. This recovery of misoperation will take place even if the MPO package is not equipped. In this case, the controlling party will be re-rung by the held party regardless of the CCDO and the recovery of misoperation options.

If an analog (500/2500 type) telephone user attempts to set up a Consultation connection by dialing a busy DN and if the Call Waiting conditions are satisfied, the controlling party will hear ringback tone and the active party will hear Call Waiting tone. If the controlling party goes on-hook before the active party has answered, the held call is disconnected regardless of the MPO options and Call Waiting tone is removed from the active party.

Multi-Party Operations – Three-Party Service

An analog (500/2500 type) telephone cannot have Call Waiting during the patience tone.

An analog (500/2500 type) telephone may be assigned both CWA and TSA Classes of Service. The user can establish a Consultation connection by answering Call Waiting during an active established call. If this is done, Control Digit features (CNFD, TGLD, and DISD) are available. Note that Programmable Control Digit TGLD, rather than a switchhook flash, is used to toggle the calls. Operation with XFA Class of Service is unchanged.

The Three-party Service feature changes the operation of Call Waiting for all analog (500/2500 type) telephones as follows (regardless of whether the sets have TSA Class of Service. If an analog (500/2500 type) telephone user activates Waiting during an active call so as to establish a Consultation connection, and if the user goes on-hook during the Consultation connection, the operation is treated as an AOCS misoperation. This recovery of misoperation will take place even if the MPO package is not equipped. In this case, the controlling party will be re-rung by the held party regardless of the CCDO and the recovery of misoperation options.

If an analog (500/2500 type) telephone user attempts to set up a Consultation connection by dialing a busy DN and if the Call Waiting conditions are satisfied, the controlling party will hear ringback tone and the active party will hear Call Waiting tone. If the controlling party goes on-hook before the active party has answered, the held call is disconnected regardless of the MPO options and Call Waiting tone is removed from the active party.

Night Restriction Classes of Service

If Call Waiting and Night Restriction for Call Waiting Class of Service (NRWA) are assigned, Call Waiting will be operational for the set only when Night Service is in effect.

Night Service Enhancements

This feature will terminate incoming Night calls to busy DNs by applying Call Waiting. This will be done even if the Night DN is an analog (500/2500 type) telephone with Call Waiting Denied (CWD) Class of Service, or if the Night DN is a Meridian 1 proprietary telephone without a Call Waiting (CWT) key assigned.

All telephones (analog (500/2500 type) and Meridian 1 proprietary) will be given Night Call Waiting tone, if the NWT prompt in LD 15 was responded to with “YES”, regardless of the Warning Tone (WTA,WTD) Class of Service setting of the set. Meridian 1 proprietary telephones will be given Night Call Waiting tone in the handset instead of the speaker buzz for Call Waiting.

On Hold on Loudspeaker

Call Waiting can be applied to a busy loudspeaker DN.

Ring Again

The user is notified that a previously busy line is free only when both the original call and the waiting call have disconnected.

Station Camp-on

Call Waiting takes precedence over Station Camp-On.

Feature packaging

Call Waiting/Internal Call Waiting is included in base system software.

For Call Waiting FFCs, the following packages are required:

- Background Terminal Facility (BGD) package 99
- Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Allow Call Waiting for analog (500/2500 type) telephones.

2 LD 11 – Allow Call Waiting for Meridian 1 proprietary telephones.

LD 10 – Allow Call Waiting for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CWD) CWA (SWD) SWA (WTA) WTD	(Deny) allow Call Waiting. (Deny) allow internal Call Waiting (if SWA is defined, CWA must also be defined). (Allow) deny warning tone.

LD 11 – Allow Call Waiting for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.

TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	SWA WTD	allow internal Call Waiting. deny warning tone.
KEY	xx CWT	Add a Call Waiting key (the M3000 must use key 24).

Feature operation

To answer a Call Waiting call on Meridian 1 proprietary telephones, follow these steps:

1 Press **Hold** when you hear a tone during a phone call.

2 Press **Call Wait** to answer the waiting call.

To return to your first call, follow these steps:

1 Press **Hold** if you want to put your second call on Hold.

2 Press the extension key that has the first call on it.

To answer a Call Waiting call on analog (500/2500 type) telephones, follow these steps:

- Flash the switchhook when you hear a beep during a phone call. Your current call is on Hold and you are connected to the waiting call.

To return to your first call:

- Flash the switchhook.

The following instructions are for using Call Waiting FFCs:

- Allow – The user must dial the Call Waiting Activate (CWGA) FFC.

- Disallow – The user must dial the Call Waiting Deactivate (ATDD) FFC or the Deactivate (DEAF) FFC.
- Use prerequisites – To set Call Waiting, the telephone must have Call Waiting Allowed (CWA) Class of Service.

Called Party Control on Internal Calls

Contents

This section contains information on the following topics:

Feature description 859
Operating parameters 860
Feature interactions 860
Feature packaging 861
Feature implementation 861
Feature operation 863

Feature description

The Called Party Control on Internal Calls (CPCI) feature allows the called party with Class of Service Malicious Call Trace Allowed to activate Malicious Call Trace (MCT) even after the calling party goes on-hook. When enabled, the calling party is prevented from releasing a call until the called party has initially released. If the calling party goes on hook, the call is put on hold and both parties are given call hold treatment. When the called party activates the call trace request, the calling party's information is printed in the call trace record.

This feature builds on the capabilities of the existing Called Party Disconnect Control (CPDC) and Malicious Call Trace (MCT) features. Called Party Control on Internal Calls is applicable on set to set calls and can be activated during or after a malicious call has occurred.

Operating parameters

The feature is applicable to all systems.

This feature is designed for stand alone application only and is not supported across a network. On a set to set call, the calling and the called party must belong to the same customer on the same system.

Called Party Control on Internal Calls (CPCI) is not supported on ISDN BRI sets.

If an Attendant Console is the calling party and involved in a call with a set, the CPCI feature functionality is not provided to the call.

The CPCI feature is not supported with features such as Attendant Recall, Override, Call Park and Privacy Release.

If the system initializes during an established call, the call remains established. Following initialization, the calling party can successfully release from a connected call prior to the called party releasing.

For this feature, the called party must have a Malicious Call Trace Allowed (MCTA) Class of Service configured in the Customer Data Block (LD 15).

The existing limitations of Malicious Call Trace apply to this feature. When MCT is requested, the information is printed in the call trace record.

If the called party does not release the connection, then the calling party remains established until the called party releases.

Feature interactions

Call Hold, Deluxe

The calling party and called party can put either party on hold. However, the calling party cannot release the call while the called party is on hold. The called party is permitted to release the call.

Held Call Clearing

With Called Party Control on Internal Call enabled, a call on hold is not cleared when the calling party releases. This occurs whether or not the Held Call Clearing feature has been activated.

Feature packaging

Called Party Control on Internal Calls requires the new package Called Party Control on Internal Calls (CPCI) package 310 and also requires Malicious Call Trace (MCT) package 107.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure Called Party Control on Internal Calls in the Customer Data Block.

2 LD 57 – Configure Malicious Call Trace Flexible Feature Code.

3 LD 10 – Enable Malicious Call Trace on Analog (500/2500 type) sets.

4 LD 11 – Enable Malicious Call Trace on Meridian 1 proprietary sets.

LD 15 – Configure Called Party Control on Internal Calls in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Change Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- MCDC	YES	Malicious Call DN/CLID printing allowed.
CPCI	YES	Called Party Control on Internal Calls allowed. NO = Called Party Control on Internal Calls denied.

LD 57 – Configure Malicious Call Trace Flexible Feature Code.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data block.
CUST	xx	Customer number, as defined in LD 15
CODE	MTRC	Malicious Call Trace Flexible Feature Code.
MTRC	xxxx	Enter Flexible Feature Code.

LD 10 – Enable Malicious Call Trace on Analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Analog Set Type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied.

LD 11 – Enable Malicious Call Trace on Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.

TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, i2002, or i2004.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied.
KEY	xxx TRC	Key Number, Malicious Call Trace.

Note: For information on the implementation of the Malicious Call Trace feature, refer to the Malicious Call Trace feature description in *Features and Services* (553-3001-306), Book 2.

Feature operation

Feature operation is described in the following scenarios.

Simple Call - Meridian 1 Proprietary Sets

Set A, the calling party, initiates a call to Set B, the called party. Set B has Class of Service Malicious Call Trace Allowed (MCTA) configured. With Called Party Control on Internal Calls (CPCI) activated, Set B can activate Malicious Call Trace even after Set A has gone on-hook. This operation is possible because Set A is not idled until Set B has gone on-hook. Depending on the type of telephone, CPCI activation is done by completing the following:

1 Enter the SPRE code and “83” access code;

2 Dial the Malicious Call Trace (MCT) Flexible Feature Code (FFC); or

3 Activate the Call Trace key.

With Called Party Control on Internal Calls, Set B can activate the Malicious Call Trace feature during an active call between Set A and Set B as well as after Set A has gone on-hook. Set A is not idled until Set B goes on-hook.

Simple Call - Analog (500/2500 type) Sets

1 Flash the switchhook. A special dial tone is heard that signifies the call is on hold.

2 Enter the SPRE code and “83” access code.

3 Dial the Malicious Call Trace Flexible Feature Code.

4 Go on-hook.

Conference Call

When the calling party and the called party both are involved in a conference call then the following operation is observed:

1 A party that called a set with CLS MCTA is only allowed to release if the called party has gone on hook and no other conferee has CLS MCTA or all remaining conferees were originators of a CPCI call.

2 In a three party conference between the called party, calling party and the recorder then the recorder is not considered as a conferee. The recorder is treated as a simple call for called party control on internal calls feature.

3 In a set to set call, if the called party establishes a conference with a trunk then the called party control on internal calls feature functionality is provided as long as the last called party (with CLS configured as MCTA) which goes on hook is a set rather than a trunk.

Transfer Call

If calling party and the called party are both involved in a simple call and either party tries to transfer the call to another set then the following operation is observed:

1 If the called party in a CPCI call attempts to transfer the call to a set with CLS MCTA, it is not allowed to complete its transfer.

2 If the called party transfers the call across the network then the called party is allowed to complete its transfer across the network and the transferred to party does not have any control on the calling party.

3 If the calling party in a CPCI call attempts to transfer the call to another set, it is not allowed to complete its transfer regardless of that set’s Class of Service.

Call on Hold

If calling party and the called party both are involved in a simple call and either party tries to put the call on hold then the following operation is observed:

1 If the calling party tries to release the call then the calling party is not allowed to release from the call.

2 If the called party tries to release the call then the called party is allowed to release from the call.

Called Party Disconnect Control

Content

This section contains information on the following topics:

Feature description 867
Operating parameters 868
Feature interactions 868
Feature packaging 869
Feature implementation 870
Feature operation 870

Feature description

Called Party Disconnect Control allows the system to control the disconnecting of calls on Central Office (CO), Foreign Exchange (FX), Common Control Switching Arrangement (CCSA), Direct Inward Dialing (DID), TIE, Wide Area telephone Service (WATS), modem, and Centralized Automatic Message Accounting (CAMA) trunks. The trunk route data block has been modified so that a route can be specified for Called Party Disconnect Control.

With Called Party Disconnect Control, an incoming trunk call answered within the system is not disconnected until the system end goes on-hook. If the calling party goes on-hook, the connection is held, allowing the call to be traced in emergency situations. If the calling party goes off-hook again, the call is not reestablished.

Operating parameters

An incoming call on a trunk route with Called Party Disconnect Control allowed can be transferred to another telephone within the system, but cannot be transferred to a trunk.

An incoming call with Called Party Disconnect Control can be forwarded to another telephone, but not to another trunk.

Tandem trunk connections are not allowed on incoming calls on trunks with Called Party Disconnect Control allowed.

If Barge-In or Busy Verify is applied to trunks with Called Party Disconnect allowed, the trunk is disconnected.

Force disconnect, through service change and maintenance, overrides Called Party Disconnect Control.

Feature interactions

Automatic Answerback

Incoming calls on a trunk with Called Party Disconnect Control allowed that terminate on a telephone with Handsfree Answerback are answered automatically. They are not disconnected automatically, however, when the calling party goes on-hook.

Conference

Trunks with Called Party Disconnect Control allowed are treated as trunks without disconnect supervision when conferenced.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Called Party Disconnect Control may not be used in the CIS market because of its signaling requirements.

Malicious Call Traced - Enhanced

Prior to this feature, the Called Party Control (CDPC) option was not supported for conference calls. The CDPC option is now supported if the conference contains exactly one recording trunk, one MCT activating party and one other trunk. This is done to make the recorder transparent to the user. The CDPC option remains unsupported for all other conference calls.

Meridian 911

The Called Party Disconnect (CPDC) feature is used to retain a 911 trunk when a 911 call is disconnected by the caller. No modification to the feature is required for Meridian 911, except lifting the CPDC and Automatic Call Distribution (ACD) NCFW limitation. 911 Calls, arriving through trunks with CPDC defined, will be allowed to NCFW, unlike non-911 ACD calls.

Meridian 911 - Call Abandon

There is no interaction with M911 Call Abandon and Called Party Disconnect Control.

Periodic Clearing Enhancement

Called Party Disconnect Control can exist on the same system and function on the same route, but is not to be used in conjunction with Periodic Clearing.

Periodic Clearing on RAN, Meridian Mail, ACD and Music

This feature enhancement is not supported if used together with Toll Operator Break-In.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 16 – Define Called Party Disconnect Control for a trunk route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer Data Block.
CUST	xx	Customer number, as defined in LD 15
CNTL	(NO) YES	(Do not) change the controls or timers.
...		
CPDC	(NO) YES	(Deny) allow Called Party Disconnect Control for the trunk route.

Feature operation

No specific operating procedures are required to use this feature.

Calling Party Name Display Denied

Content

This section contains information on the following topics:

Feature description 871
Operating parameters 873
Feature interactions 874
Feature packaging 875
Feature implementation 876
Feature operation 877

Feature description

This enhancement to the Calling Party Name Display feature allows a customer to define, on an originating set, whether or not to allow the display of the calling and called party name and/or digits on the terminating set. This option can be defined individually for each customer set, and applies to all Meridian telephone types. The display of digits is controlled by “digit display allowed on other set” (DIGA) or “digit display denied on other set” (DIGD) Class of Service. The name display is controlled by the “name display allowed on other set” (NAMA) or “name display denied on other set” (NAMD) Class of Service.

The following scenarios are possible, where set A is the originating set and set B is the terminating set. DIGA "Allowed" and "Denied" indicates whether or not the called party digits display are allowed or denied on the set. If the display of digits is denied, the digits are replaced by four dashes (for an internal call) or seven dashes (for an external call). If the name display is denied (that is, NAMD), the name is replaced by "XXXX".

In the following example, originating set A has DIGA and NAMA Class of Service, and terminating set B also has DIGA and NAMA Class of Service. During an established call, the respective displays would be as follows:

Figure 11
Display of Calling Party Denied example

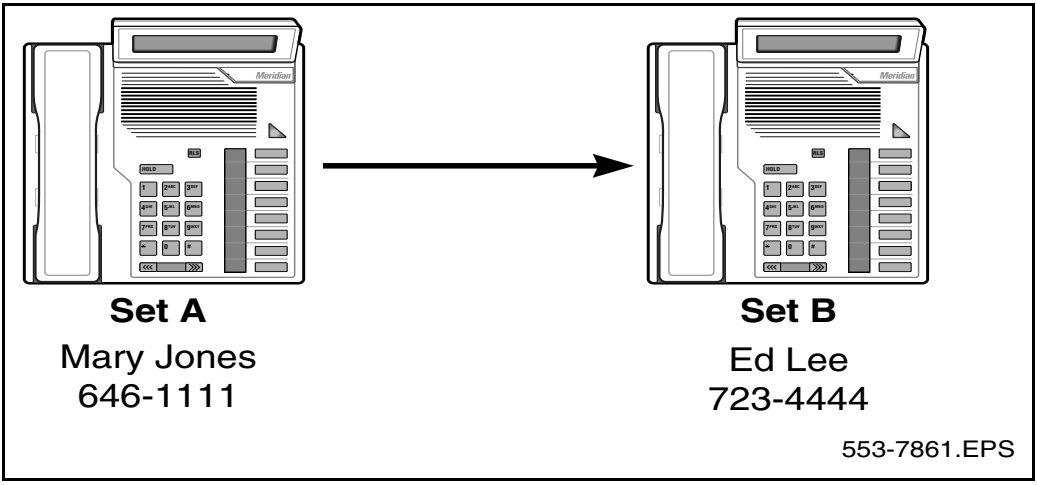
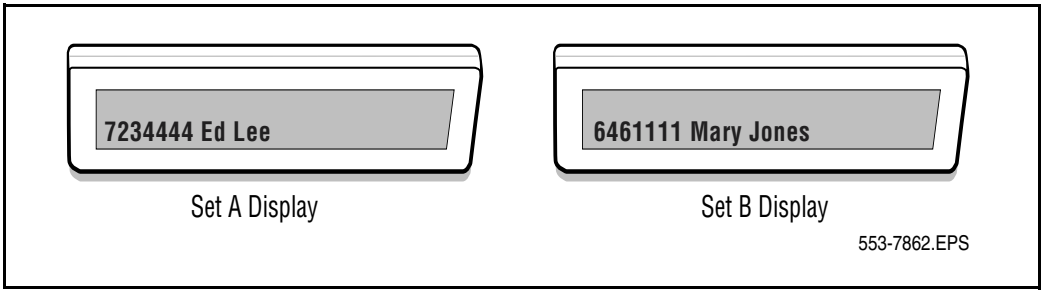
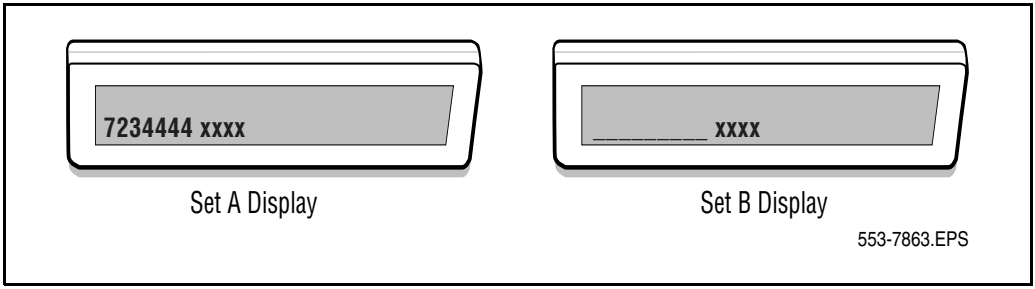


Figure 12
Example of displays when the originating and terminating sets both have Class of Service set to DIGA and NAMA



If set A has DIGD and NAMD Class of Service, and set B also has DIGD and NAMD Class of Service, the displays would be as follows (keep in mind that set A displays the dialed digits even though set B has DIGD Class of Service):

Figure 13
Example of displays when the originating and terminating sets both have Class of Service set to DIGD and NAMD



Operating parameters

This enhancement pertains to both standalone and network environments.

The called party digits are displayed on the originating set, regardless of the Class of Service of the terminating set.

Feature interactions

Attendant Consoles

The Calling Party Name Display Denied enhancement cannot be applied to Attendant Consoles.

Call Forward
Call Transfer

During a Call Forward or Call Transfer, the calling party digits and forwarding/transferring party digits are displayed on the terminating set. This display is allowed or denied depending on the Class of Service of the calling set and the forwarding/transferring set. The name of the forwarding/ transferring set is not displayed on the calling and terminating set.

Calling Party Privacy

For outgoing calls, if the Calling Party Privacy (CPP) package is equipped, the CPP feature will take precedence over the Calling Party Name Display Denied feature for restricting the Calling Party Name and Number. For example, if an outgoing ISDN call is marked as a CPP call, the outgoing SETUP message will include the Calling Party Number IE with the Presentation Indicator set to “Presentation Restricted” and the Display IE with the CPND Indicator set to “Presentation Denied”, to inhibit both the Calling Party Number and Name being displayed on the terminating set, regardless of whether or not the Calling Party Name Display Denied feature allows the display of the Calling Party Name and/or Number.

The Calling Party Name Display Denied feature takes precedence over the CPP feature for displaying an incoming ISDN call. If International Supplementary Features (SUPP) package 131 is equipped, an incoming ISDN call with the Presentation Indicator set to “Presentation Restricted” in the Calling Party Number IE or the CPND Indicator set to “Presentation Denied” in the Display IE will be marked as a CPP call, and will display “ACOD + Member” or “XXXX” as for the Calling Party Name Display Denied feature.

Conference

Calling Party Name Display Denied does not apply to conference calls.

ISDN QSIG Name Display

Call Party Name Display and Calling Party Name Display Denied interact with ISDN Q Interface Signaling Protocol (QSIG) Name Display, depending on the Name Display configuration in LD 16 for BRI or LD 17 for PRI. When a QSIG network is interacting with an Meridian Customer Defined Network (MCDN) network providing network capability ND3, both the MCDN and QSIG Name Display feature function on the same level.

Multiple Appearance Directory Numbers

For a ringing call to a Multiple Appearance Directory Number (DN), the name on the calling set display can be suppressed by configuring any of the Terminal Numbers with NAMD Class of Service. The digit display on the calling set cannot be suppressed—the called digits are displayed even though the Class of Service on any of the Terminal Numbers is DIGD. The called set display is subject to the Class of Service of the calling party. For an established call to a Multiple Appearance Directory Number (DN), the calling set display is subject to the Class of Service configured for the answering set. The answering set display only is subject to the Class of Service of the calling party—the displays of the other sets in the Multiple-appearance group are blank.

Office Data Administration System (ODAS)

Depending on the Class of Service of the originating set, the ODAS designator is displayed or replaced by Xs, up to the maximum number of characters that the designator may have.

Feature packaging

Calling Party Name Display Denied requires International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 10 – Configure the Calling Party Name Denied Class of Service for analog (500/2500 type) telephones.
2. LD 11 – Configure the Calling Party Name Display Denied Class of Service for proprietary telephones.

LD 10 – Configure the Calling Party Name Denied Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Telephone type.
...		
CLS	(DDGA) DDGD (NAMA) NAMD	(Allow) deny DN to be displayed on other set. (Allow) deny name to displayed on other set.

LD 11 – Configure the Calling Party Name Display Denied Class of Service for proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Telephone type.
...		
CLS	(DDGA) DDGD (NAMA) NAMD	(Allow) deny DN to be displayed on other set. (Allow) deny name to be displayed on other set.

Feature operation

No specific operating procedures are required to use this feature.

Calling Party Privacy

The Calling Party Privacy (CPP) feature enables the system to support the blocking of a Calling Party's Number and Name from being displayed at the terminating set on an individual call basis. Users can dial a Calling Party Privacy code (for example, *67 from a proprietary set or 1167 from an analog (500/2500 type) set) to prevent their telephone number and name from being displayed on a receiving telephone across the Public Switched Telephone Network (PSTN). Internal calls within the system have originating numbers or names displayed, even though the originating call has requested privacy.

This feature also allows a per-line blocking Class of Service to be programmed for station sets for public network calls. This relieves the user from having to dial the Flexible Feature Code (FFC) for every call, but in every other way is equivalent to the per-call blocking.

Please refer to *ISDN Primary Rate Interface: Features* (553-3001-369) for complete information.

Calling Party Privacy Override

Content

This section contains information on the following topics:

Feature description 881
Operating parameters 884
Feature interactions 886
Feature packaging 896
Feature implementation 897
Feature operation 901

Feature description

Calling Party Privacy Override (CPPO) enhances the functionality of the Calling Party Privacy (CPP) feature. With Calling Party Privacy Override, calling party information can be selectively unblocked on a per-call basis.

With the Calling Party Privacy Override feature, a circuit switched network user can selectively unblock calling party information on a per-call basis when Class of Service is set to CLBA. The user unblocks the calling party information by dialing a Calling Party Privacy Override Flexible Feature Code prior to dialing the destination number.When the CPPO Flexible Feature Code is dialed before the destination number, the user's calling party information is displayed on the terminating set. The default for the Calling Party Privacy Override Flexible Feature Code is "*82" for proprietary sets and "1182" for analog (500/2500 type) sets. The Calling Party Privacy Override Flexible Feature Code is defined in Overlay 57.

CPPO is provisioned on a trunk route basis. Any trunk type that can support an outgoing call can request the CPPO feature (see "Operating parameters" on page 884 for more information).

When the CPPO Flexible Feature Code is dialed prior to the normal dialing sequence, the call is marked as a CPPO call. The CPPO Flexible Feature Code is then removed from the dialed digits stored in the call register. If the outgoing trunk route provisions CPPO, then the Privacy Override Indicator is sent to the far end, and the Calling Party Number and Name information is displayed on the receiving telephone. If the outgoing trunk route does not provision CPPO, the call does not carry the Privacy Override Indicator.

The following example illustrates Calling Party Privacy Override functionality:

1 Set A, a Meridian 1 proprietary set with Class of Service set to CLBA, goes off-hook.

2 Set A dials the Calling Party Privacy Override Flexible Feature Code, defined in Overlay 57. Calling Party Privacy Override is initiated.

3 Set A dials the destination number for Set B.

4 The call rings on Set B.

5 The calling party information of Set A is presented on the display screen of Set B.

Outgoing calls

For an outgoing non-ISDN trunk call, the Privacy Override Indicator is defined on the outgoing trunk route. The CPPO Flexible Feature Code is outpulsed to the far end provided that the outgoing trunk route provisions CPPO. If CPPO is not provisioned on the trunk route, then the call does not carry the Privacy Override Indicator.

For an outgoing ISDN call from one system to another, the Privacy Override Indicator is represented when the Presentation Indicator field is set to "Presentation Allowed" in the Calling Party Number Information Element (IE) and the Call Party Name Display (CPND) Indicator field is set to "Presentation Allowed" in the Display IE.

For an outgoing ISDN call to the Central Office, the Privacy Override Indicator is represented when the Presentation Indicator field is set to "Presentation Allowed" in the Calling Party Number IE and when the CPND information is included in the Display IE.

Incoming calls

An incoming ISDN call is recognized as a CPPO call (that is, it carries the Privacy Override Indicator) if the Presentation Indicator field is set to "Presentation Allowed" in the Calling Party Number IE and if the CPND Indicator is set to "Presentation Allowed" in the Display IE (if it exists).

When an incoming call is on a non-ISDN route, the system does not receive the Privacy Override Indicator.

Tandem Calls

Incoming ISDN calls

ISDN to ISDN tandem

For an incoming call tandeming through the system, any incoming Privacy Override Indicator is only repeated to the outgoing trunk route that also has CPPO provisioned.

When an incoming ISDN trunk call is tandemed through an ISDN trunk to a system, the Presentation Indicator or the CPND Indicator, received from the incoming ISDN trunk, is tandemed to the outgoing ISDN trunk.

When an incoming ISDN trunk call is tandemed through an ISDN trunk to a CO, the Presentation Indicator received from the incoming ISDN trunk is tandemed to the outgoing ISDN trunk. If the Display IE with the CPND Indicator set to "Presentation Allowed" is received from an incoming ISDN trunk, the Display IE, containing the Call Party Name, is sent across in the SETUP message tandemed to the outgoing ISDN trunk.

ISDN to non-ISDN tandem

When an incoming ISDN trunk call is tandemed to a non-ISDN trunk, the incoming call is treated as a CPPO call only if both the CLID and CPND Indicators are set to "Allowed". Otherwise, the call is treated as a CPP call.

Incoming non-ISDN calls

For incoming non-ISDN calls, the system does not receive the Privacy Override Indicator.

When a call on an incoming non-ISDN route is tandemed on the system, the call is tandemed based on how the CPP flag (TCPP) prompt is defined in the Route Data Block for the outgoing route.

When TCPP is set to YES, an incoming non-ISDN call tandemed to this route is treated as a CPP call.

When TCPP is set to NO, an incoming non-ISDN call tandemed to this route is treated as a CPPO call.

Non-ISDN to ISDN tandem

Even though a Privacy Override Indicator is not provided for an incoming non-ISDN trunk call, if the outgoing route has TCPP set to NO, the Presentation Indicator field in the Calling Party IE is set to “Presentation Allowed”.

Non-ISDN to non-ISDN tandem

A Privacy Override Indicator is not provided for an incoming non-ISDN trunk call. If the outgoing route has TCPP set to NO, the Privacy Override Indicator defined for that route is outpulsed, provided that the outgoing route provisions CPPO.

Operating parameters

Central Office Trunks (COT), Foreign Exchange (FEX), Wide Area Telephone Service (WATS), and Direct Inward Dial (DID) are the only trunk route types (including ISA service routes) that can outpulse the Privacy Override Indicator for an outgoing non-ISDN call. All ISDN trunk routes provision the CPPO feature.

A non-ISDN trunk route does not provision the CPPO feature if the Outpulse Asterisk and Octothorpe (OPAO) package (package 104) is configured. During SYSLOAD, the CPPO database is removed from the non-ISDN trunk routes if the OPAO package is configured.

The Privacy Override Indicator, defined for a non-ISDN trunk route (dial-pulse or digitone), consists of any four arbitrary digits from 0-9. The asterisk (*) or octothorpe (#) cannot be part of the Privacy Override Indicator for dial-pulse trunks. For digitone trunks, the asterisk (*) can only be the first digit of the Privacy Override Indicator Flexible Feature Code.

The asterisk and octothorpe are not outpulsed if the OPAO package is configured. The asterisk signals a 3-second pause and the octothorpe indicates end-of-dialing. The octothorpe cannot be used in a Privacy Override Indicator.

Privacy Override Indicators are not received from the CO or non-ISDN DID trunks.

The CPPO Flexible Feature Code cannot conflict with any internal DN, including the Special Prefix (SPRE) code.

When a user dials the Flexible Feature Code defined for the CPPO feature and if CPPO is not provisioned on the outgoing trunk route, the call proceeds without carrying the Privacy Override Indicator.

The CPPO feature does not affect whether or not the Calling Party Number and Name information is displayed for internal calls within the system, even if the originator requests CPPO.

All incoming non-ISDN calls with the Privacy Override Indicator terminate on the system. If the Privacy Override Indicator is not defined in the Flexible Feature Code for CPPO, an overflow tone (unrecognized digits) is provided to the user.

If the Stored Number Redial (SNR)/Last Number Redial (LNR) feature is used by the originator of a CPPO call to store the dialed digits, the CPPO Flexible Feature Code is stored against the SNR/LNR database. If the user removes that CPPO Flexible Feature Code and then the SNR/LNR feature is used to re-initiate the call, overflow tone is returned to the user.

ISDN implementation for this feature includes DMS100/250, SL-100, AT&T4, AT&T5, TR-1268 (NI-2), Meridian Customer Defined Network (MCDN) Private Networks, EuroISDN, QSIG, and BRI trunks.

The CPPO feature is supported on the following International PRI (IPRI) connectivities: Ericsson AXE-10 CO Connectivity (Australia), Ericsson AXE10-CO Connectivity (Sweden), French Numeris CO Connectivity, Japan D70 CO Connectivity, Swissnet 2 CO Connectivity, SYS-12 CO Connectivity, 1TR6 CO Connectivity (Germany), and Asia Pacific ISDN Phase 2.

The CPPO feature supports the following North American connectivities: DMS100/250, S1100, Lucent #4 ESS (ESS4), Lucent #5 EES (ESS5), and TR-1268 (NI-2).

CPPO does not support R2MFC signaling.

Feature interactions

Attendant Consoles

A CPPO call can be originated from any Meridian 1 Attendant Console. Attendant Consoles request CPPO by preceding the normal dialing sequence with the Flexible Feature Code for CPPO.

Attendant Consoles can also initiate a CPPO call using the Autoline key. An outgoing trunk call, initiated by pressing the Autoline key, carries the Privacy Override Indicator if the CPPO Flexible Feature Code, followed by the normal dialing sequence, is stored against the Autoline key. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) stored against the Autoline key.

The CPPO Flexible Feature Code can also be stored against the Autoline key. An outgoing CPPO call can then be initiated by pressing the Autoline key followed by manually dialing the destination number.

An outgoing CPPO call can also be initiated by dialing the CPPO Flexible Feature Code followed by pressing the Autoline key, on which the normal dialing sequence of digits for the destination number is stored.

Autodial

An outgoing trunk call, initiated by pressing the Autodial key, carries the Privacy Override Indicator if the CPPO Flexible Feature Code followed by the normal dialing sequence is stored against the Autodial key. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) stored against the Autodial key.

The CPPO Flexible Feature Code can be stored against the Autodial key. In this case, an outgoing CPPO call can be initiated by pressing the Autodial key followed by manually dialing the normal sequence of digits for the destination number.

An outgoing CPPO call can also be initiated by dialing the CPPO Flexible Feature Code followed by pressing the Autodial key on which the normal dialing sequence of digits for the destination number is stored.

Automatic Call Distribution

Calls placed by means of Enhanced Automatic Call Distribution (ACD) Routing, Enhanced Interflow, Enhanced Night Call Forward, Enhanced Network Routing, and Network ACD recognize the originator's CPPO request.

Automatic Call Distribution MAX

If the CPP package is equipped, ACD MAX reports include the Calling Line Identification (CLID) for incoming ISDN calls that have the CLID Presentation Indicator set to "Allowed".

Basic Rate Interface

Although Basic Rate Interface (BRI) networking is not supported in North America, CPPO treats BRI trunk calls in the same manner as an ISDN trunk call.

Call Detail Recording

Call Detail Recording (CDR) records continue to include the Calling Party Number even if the caller has requested CPPO. When the CDR record is generated, the CPPO Flexible Feature Code dialed by the originator is included in the DIGIT field (if it displays the dialed digits).

The CPPO Flexible Feature Code dialed by the originator is not included in the DIGIT field if it displays the outpulsed digits. The Privacy Override Indicator, outpulsed by an outgoing non-ISDN trunk route that provisions CPPO, is included in the outpulsed digits.

Call Pickup Network Wide

When an incoming trunk call with the Privacy Override Indicator is picked up by a remote set (the requesting party), the Calling Party Number and Name is displayed on the requesting set.

Call Hold

When an incoming trunk call with the Privacy Override Indicator is taken off hold, the Calling Party Number and Name information is displayed on the set.

Call Forward All Types
Hunt
Network Hunt

The existing call redirection functionality is not changed by this feature.

When an incoming ISDN trunk call with the Privacy Override Indicator is forwarded into the public or private networks, the Privacy Override Indicator is tandemed to the far end to allow the display of the Calling Party Number and Name, provided that the outgoing trunk route on the tandem node has CPPO provisioned.

When an incoming ISDN call with Calling Party Number and Name set to "Presentation Allowed" is forwarded to a set within the same node, the Calling Party Number and Name is displayed on the terminating set.

When an incoming non-ISDN trunk call is forwarded onto a trunk, the outgoing trunk call from the tandem node carries the Privacy Override Indicator, provided that the outgoing trunk route on the tandem node has CPPO provisioned. Also, the TCPP prompt in the Route Data Block must be set to NO.

The CPPO Flexible Feature Code can be stored on the forwarding Directory Number (DN), including the forwarding DN for Call Forward All Calls, Hunt DN and Flexible Call Forward No Answer DN (FDN).

If CPPO is requested on the forwarding DN and the call is forwarded across an ISDN link, the outgoing SETUP message includes the Redirecting Number IE (if it exists) with the Presentation Indicator set to “Presentation Allowed”.

If CPPO is requested on the forwarding DN and the call is forwarded across a non-ISDN link, no Privacy Override Indicator is outpulsed to the terminating node if the originating set did not request CPPO. This is because no Redirecting Number information is sent across a non-ISDN link.

When an internal call is forwarded into the public or private networks, if the originator requests CPPO and the outgoing trunk route provisions CPPO, the Privacy Override Indicator is sent to the far end to allow the display of the Calling Party Number and Name.

Call Pickup

With CPPO activated, when an incoming trunk call with the Privacy Override Indicator is picked up locally, the Calling Party Number and Name information is displayed on the terminating set.

Call Transfer

As per existing operation, if an incoming non-ISDN call is transferred or an incoming ISDN call is transferred to a non-ISDN trunk, the Connect Party Number and Name information is not passed to the terminating node. The CPPO feature does not change this operation.

When an incoming call with the Privacy Override Indicator is transferred across the MCDN network or to a local set, the originator’s calling party information is displayed on the final terminating set.

Calling Line Identification Restriction

Basic Rate Interface (BRI) sets do not support the Flexible Feature Code (FFC) feature. CPPO can only be requested by applying the existing Calling Line Identification Restriction (CLIR) Service option. This is done by setting the soft key "ID PRES" (if it exists) to "Allowed" or the Presentation of CLID to far end on outgoing calls (PRES) prompt to YES in Overlay 27. Then an outgoing ISDN/non-ISDN trunk call carries the Privacy Override Indicator if the outgoing trunk route provisions CPPO. However, if the Calling Party Number Information Element (IE) with the Presentation Indicator set to "Presentation Denied" is included in the SETUP message generated by the Basic Rate Interface (BRI) terminal, then the BRI terminal does not allow CPPO. This is because the Presentation Indicator, generated by the BRI terminal, always overwrites the Calling Line Identification Restriction (CLIR) service option.

Calling Party Privacy

If the user requests both Calling Party Privacy and Calling Party Privacy Override, then the feature last requested takes precedence. The Flexible Feature Code dialed last determines the type of call.

If a set with Class of Service set to CLBA requests CPPO by dialing the CPPO Flexible Feature Code, then the call is treated as a CPPO call. If a set with Class of Service set to CLBD requests CPP by dialing the CPP Flexible Feature Code, then the call is treated as a CPP call.

If a user dials the Flexible Feature Code for CPPO followed by the Flexible Feature Code for CPP, then the call is treated as a CPP call. If a user dials the Flexible Feature Code for CPP followed by the Flexible Feature Code for CPPO, then the call is treated as a CPPO call.

Calling Party Privacy and Call Forward

Set A, requesting CPPO, calls Set B. Set B Call Forwards All Calls to Set C. The CPP Flexible Feature Code is part of the forwarding DN. Set A's number and name is displayed on Set C as the Calling Party Number and Name; although, no redirecting number is displayed on Set C. The tandem node sends the Display IE with the Presentation Indicator set to "Allowed" and the Redirecting Number IE with the Presentation Indicator set to "Restricted".

Set A, requesting CPP, calls Set B. Set B Call Forwards All Calls to Set C. The CPPO Flexible Feature Code is part of the forwarding DN. Set B's number is displayed on Set C as the Redirecting Number; although, no Calling Party Number and Name is displayed on Set C. The tandem node sends the display IE with the Presentation Indicator set to "Restricted" and the Redirecting Number IE with the Presentation Indicator set to "Allowed".

Calling Party Privacy and Call Transfer

Set A, requesting CPPO, calls Set B. Set B answers the call, requests CPP, and initiates a transfer to Set D. After the transfer is complete, Set A's Calling Party Number and Name is displayed on Set D. The request made by the connected party takes precedence over the transferring party while displaying the Connect Party Number and Name.

Set A, requesting CPP, calls Set B. Set B answers the call, requests CPPO, and initiates a transfer to Set D. After the transfer is complete, Set A's Calling Party Number and Name is not displayed on Set D. The request made by the connected party takes precedence over the transferring party while displaying the Connect Party Number and Name.

Conference

The CPPO feature passes the Privacy Override Indicator to the terminating set in order to display the Calling Party Number and Name, if the Conference feature is used for the purpose of performing a transfer.

Display of Calling Party Denied

When the CPP package is equipped, the CPPO feature takes precedence over the Display of Calling Party Denied (DPD) feature for allowing the Calling Party Number and Name to be displayed. For example, when an outgoing ISDN call is marked as a CPPO call, then the outgoing SETUP message includes the Calling Party Number IE with the Presentation Indicator set to "Presentation Allowed" and the Display IE with the CPND Indicator set to "Presentation Allowed". This enables both the Calling Party Number and Name to be displayed on the terminating set, regardless of whether the DPD feature allows or denies the display of the Calling Party Number and/or Name.

E.164 ESN Numbering Plan Enhancement

CPPO can be requested for ESN calls by preceding the dialing sequence with the Flexible Feature Code defined for the CPPO feature. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed for the destination DN.

Feature Group D

When an incoming Feature Group D (FGD) call terminates at a system locally, the received 10-digit Automatic Number Identification (ANI) is displayed on the terminating set if the Show ANI Digits on Terminal Displays (SHAN) field is set to YES in the FGD data block that is associated with the incoming trunk route. If the originator requests CPPO, the end office sends the 10-digit ANI to the circuit switched network.

If an incoming FGD call is routed to another switch through ISDN Primary Rate Interface (PRI) or ISDN Signaling Link (ISL), the outgoing SETUP message includes the 10-digit ANI (if it exists) as the Calling Party Number (CLID) with the Presentation Indicator set to “Presentation Allowed”. This occurs if the incoming call requests CPPO. CPPO takes precedence over the SHAN field that is defined in the FGD data block and is associated with the incoming trunk route to allow the 10-digit ANI display.

Hot Line

Hot line calls carry the Privacy Override Indicator if the CPPO Flexible Feature Code followed by the normal dialing sequence is stored in the Hot Line DN. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed for the Hot Line DN.

Last Number Redial

The Last Number Redial (LNR) feature stores the CPPO Flexible Feature Code in the LNR database if the CPPO Flexible Feature Code was included in the last number dialed by the user. The outgoing redialed calls also send the Privacy Override Indicator to the far end.

Incoming Trunk Programmable Calling Line Identification

When the incoming trunk route is a non-ISDN route, the billing number (CLID) assigned by the incoming trunk route is passed to the CO with the Presentation Indicator field set to “Presentation Allowed”, if the outgoing ISDN trunk route has the TCPP prompt set to NO.

When the incoming trunk route is an ISDN route, the “Allowed” Presentation Indicator is tandemed to the outgoing trunk route. If the Presentation Indicator is set to “Presentation Allowed” or no Calling Party Number IE is received on the incoming trunk route, the billing number assigned by the incoming trunk route is passed to the CO with the Presentation Indicator field set to “Presentation Allowed”, if the incoming trunk route has the Billing Number Display (BDSP) prompt set to YES or NO.

ISDN Signaling Link

CPPO treats an ISDN Signaling Link (ISL) call in the same manner as an ISDN trunk call.

Malicious Call Trace

An incoming call to a set with the Malicious Call Trace (MCT) feature activated includes the Terminal Number (TN) of the calling party in the MCT record, whether or not the caller has requested CPPO.

Meridian 911

An incoming 911 call with Automatic Number Identification (ANI) information always displays the ANI digits on the terminating set or passes the ANI information to the Meridian 911.

Meridian Interactive Voice Response

An incoming ISDN call with the CLID Presentation Indicator set to “Allowed” sends the CLID to the Meridian Interactive Voice Response (IVR) if the CPP package is equipped.

Meridian Link

If the CPP package is equipped, an incoming ISDN call with the CLID Presentation Indicator set to “Allowed” includes the CLID in the Application Module Link (AML) messages sent to the Meridian Link module.

Meridian Mail

When an incoming ISDN call with the CLID Presentation Indicator set to "Allowed" terminates on Meridian Mail, the CLID passed to Meridian Mail is recorded. The call is treated by Meridian Mail as an external call.

Calls placed by means of Through Dial can request Calling Party Privacy Override. These calls involve the person accessing Meridian Mail (mailbox user or incoming caller) dialing 0 followed by any telephone number. The caller is able to dial a CPPO Flexible Feature Code plus the normal dialing sequence, following the 0. The asterisk (*) or octothorpe (#), as part of the CPPO Flexible Feature Code, are rejected by Meridian Mail. Therefore, the CPPO Flexible Feature Code can only consist of seven digits (0-9).

Meridian MAX

If the CPP package is equipped, an incoming ISDN call with the CLID Presentation Indicator set to "Allowed" sends the CLID to Meridian MAX.

Network Call Redirection

If a set receives a call and is then redirected to the public network on an ISDN trunk that supports call redirection, then the redirecting IE in the outgoing SETUP message has the Presentation Indicator set accordingly. For instance, if the call that had requested CPPO is redirected, the outgoing SETUP message has the Presentation Indicator set to "Allowed".

Network Message Center

An incoming trunk call with the Privacy Override Indicator displays the Calling Party Number and Name on the Message Center operator's terminal.

Network Ring Again

A call placed by means of the Network Ring Again feature recognizes the CPPO request from when the call was originally dialed.

Nortel Symposium Call Center

As per existing operation, an incoming CPPO call routed to Nortel Symposium Call Center contains the CLID.

Private Line Service

The Private Line Service feature outpulses the Privacy Override Indicator only if it is dialed by the originator. The asterisk (*) is outpulsed to the far end only if it is an Outpulse Asterisk and Octothorpe (OPAO) call. Otherwise, the asterisk (*) signals a three-second pause.

Remote Virtual Queuing

The Remote Virtual Queuing feature has automatic re-try capabilities that are used when congestion is encountered within the network. The same Calling Party Privacy Override considerations are provided to the “re-tries” as were provided to the originally dialed call.

Ring Again – Busy Trunk

A call that is automatically redialed by the Ring Again - Busy Trunk feature recognizes the CPPO requested when the call is originally dialed.

Speed Call
System Speed Call

When an outgoing trunk call is initiated by dialing a Speed Call code, the Speed Call code carries the Privacy Override Indicator if the CPPO Flexible Feature Code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

The user can also store the CPPO Flexible Feature Code in the Speed Call Entry (or Speed Call key). An outgoing CPPO call can be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

Stored Number Redial

In the Stored Number Redial (SNR) programming mode, the user can store the CPPO Flexible Feature Code, followed by the normal dialing sequence in the SNR database. The outgoing calls originated by the Stored Number Redial feature send the Privacy Override Indicator to the far end. The CPPO Flexible Feature Code is counted against the maximum number of digits (currently 31) allowed by the SNR feature.

During an active call on a Meridian 1 proprietary set, the Stored Number Redial feature stores the CPPO Flexible Feature Code in the SNR database if the CPPO Flexible Feature Code is included in the number dialed by the originator. The outgoing redialed calls also send the Privacy Override Indicator to the far end.

Trunk Anti-Tromboning

When trunks are removed, due to the Trunk Anti-Tromboning (TAT) operation, an ISDN call recognizes the CPPO/CPP requested by the originator.

Trunk Optimization Before Answer

An optimized call, due to Trunk Optimization Before Answer (TRO) operation, recognizes the CPPO/CPP requested by the originator.

Virtual Network Services

CPPO treats Virtual Network Services (VNS) trunk calls in the same manner as ISDN trunk calls. For instance, CPPO does not affect the existing VNS operation. If CPPO was requested when originating a call, the Presentation Indicator field of CLID is set to “Presentation Allowed”.

VISIT

The VISIT which connects to a set receives the Calling Party Number or Name, since an incoming CPPO call sends the Calling Party Number or Name to the set for display.

Feature packaging

The Calling Party Privacy Override feature requires the following package:

- Calling Party Privacy (CPP) package 301, which has the following dependency:
  - — Flexible Feature Codes (FFC) package 139.

For Calling Party Name Display, Calling Party Name Display (CPND) package 95 is required. ISDN package 145 is required for ISDN routes.

Note: Non-ISDN trunks must restrict the Outpulse Asterisk and Octothorpe (OPAO) package 104 to provision the Calling Party Privacy Override feature.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Configure Privacy Override Indicators for a Non-ISDN route. Configuration procedures require that the following conditions are met:

2 LD 16 – For an ISDN trunk route, set the TCPP flag in RDB to tandem non-ISDN calls on to this route. Configuration procedures require that the following conditions are met:

3 LD 57 – Define the Flexible Feature Code for the Calling Party Privacy Override feature.

4 LD 10/11 – Activate Calling Party Number and Name per-line blocking.

LD 16 – Configure Privacy Override Indicators for a Non-ISDN route. Configuration procedures require that the following conditions are met:

- CPPO is configurable on COT, DID, FEX, WAT and ISA routes.
- OAPO package 104 is restricted or unequipped.
- Route is either OGT (outgoing) or IAO (incoming and outgoing).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15

ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
...		
CPP	 YES	Calling Party Privacy/Privacy Override (CPP/CPPO) flag. Enable CPP/CPPO feature and configure parameters. (NO) = CPP/CPPO feature is disabled is the default.
- TCPP	 (NO) YES	CPP/CPPO flag treatment for an incoming non-ISDN trunk call tandemed to this trunk route. Outgoing call will carry the Privacy Override Indicator (default). Outgoing call will carry the Privacy Indicator.
- DTPI	(*67) nnnn	Digitone Trunk Privacy Indicator nnnn = 0-9999, an asterisk (*) can be entered as the first digit.
- DPPI	0-(1167)-9999	Dial-pulse Trunk Privacy Indicator
- DTPO	(*82) nnnn	Digitone Trunk Privacy Indicator nnnn = 0-9999, an asterisk (*) can be entered as the first digit.
- DPPO	0-(1182)-9999	Dial-pulse Trunk Privacy Indicator

LD 16 – For an ISDN trunk route, set the TCPP flag in RDB to tandem non-ISDN calls on to this route. Configuration procedures require that the following conditions are met:

- The CPP package 301 is equipped.
- Route is either OGT (outgoing) or IAO (incoming and outgoing).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15

ROUT		Route number
	0-511 0-127	For Large Systems For Small Systems and Succession 1000 systems
		Note: All ISDN trunk routes are CPPO configurable.
...		
CPP		Calling Party Privacy/Privacy Override (CPP/CPPO) flag.
	YES	Enable CPP/CPPO feature and configure parameters. (NO) = CPP/CPPO feature is disabled is the default.
- TCPP		CPP/CPPO flag treatment for an incoming non-ISDN trunk call tandemed to this trunk route.
	(NO)	Outgoing call will carry the Privacy Override Indicator (default).
	YES	Outgoing call will carry the Privacy Indicator.

LD 57 – Define the Flexible Feature Code for the Calling Party Privacy Override feature.

Prompt	**Response**	**Description**
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15.
FFCT	(NO) YES	Flexible Feature Confirmation Tone denied. Flexible Feature Confirmation Tone allowed.
...		
CODE	CPP	CPP Flexible Feature Code
- CPP	xxxx	Calling Party Privacy code xxxx = 0-9999, an asterisk (*) can be entered as the first digit. The Flexible Feature Code may be up to 4 digits, or up to 7 digits with the Directory Number Expansion (DNXP) package (150).
- CPP	xxxx	Change the CPP code or enter a <CR> to accept.
CODE	CPPO	CPPO Flexible Feature Code

- CPPO	xxxx	Calling Party Privacy Override code xxxx = 0-9999, an asterisk (*) can be entered as the first digit. The Flexible Feature Code may be up to 4 digits, or up to 7 digits with the Directory Number Expansion (DNXP) package (150).
- CPPO	xxxx	Change the CPPO code or enter a <CR> to accept.

LD 10/11 – Activate Calling Party Number and Name per-line blocking.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Type of set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	d...d	Designator The response d...d represents an Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number, as defined in LD 15
...		
CLS	CLBA	Activate Calling Party Number and Name per-line blocking. CLBD = Deactivate Calling Party Number and Name per-line blocking (default).

Feature operation

For a user to override the Calling Party Number and Name per-line blocking allowed (CLBA) Class of Service, the following steps must be performed.

1 The user goes off hook.

2 The user initiates a call by dialing the Calling Party Privacy Override Flexible Feature Code, defined in LD 57.

3 The user dials the destination number.

Camp-On to a Set in Ringback or Dialing

Contents

This section contains information on the following topics:

Feature description 903
Operating parameters 904
Feature interactions 905
Feature packaging 905
Feature implementation 906
Feature operation 906

Feature description

This feature allows a station or attendant to Camp-on an external trunk call to another station that is in a ringback or dialing state. If the station on which the call is camped on becomes idle without going into established state, the camped-on call rings the station automatically.

This capability applies to standalone and networking environments. Within a networking environment, the station affecting the Camp-on and the desired party can be anywhere in the network.

If the Flexible Tones and Cadences package is equipped and periodic Camp-on tones and cadences are defined, periodic Camp-on tone is given to the desired station when it goes into established state indicating that a call is camped on. For analog sets, this is in the form of a tone. For digital sets, it is a periodic buzz. The Camp-on tone lasts for the duration of the Camp-on. The desired station receives periodic Camp-on tone, if the station has Warning Tone Allowed class of service and the customer has the Camp-on Tone Allowed option. Music is provided to the camped-on station, if the Music package is equipped and defined for the customer.

During Camp-on, as soon as the attendant releases the call or the station completes the transfer, recall timing begins (the Recall Timer is configured in LD 15). If the timer times out, the Camp-on is recalled to the attendant. If the attendant is busy, the recall is queued against the attendant. The call can no longer be camped on to the desired station without affecting another Camp-on. If the attendant is in Night Service, the Camp-on receives night treatment.

If the desired party is on a different node, and Network Attendant Service (NAS) is equipped, the Camp-on is routed according to the NAS routing table. If the Camp-on is recalled to the local attendant, and the local attendant is busy, the recall is queued to the attendant. During this time, the call may still be answered by the desired station (the call remains camped on until the attendant answers the recall). This capability is that of the Slow Camp-on Recall.

Operating parameters

Only one call at a time can be camped on to a station in dialing or ringback state.

The cadence for Periodic Camp-on can be defined in LD 56. Periodic Camp-on can be allowed or denied on a customer and set basis.

Feature interactions

Attendant Forward No Answer

Camp-on recall takes precedence over the Attendant Forward No Answer recall. However, if during the recall the customer goes into Night Service and the recall is not answered by the night DN, the call is disconnected according to the Attendant No Answer feature processing.

Call Forward All Calls
Call Waiting

Call Waiting and Call Forward All Calls take precedence over Camp-on.

First-Second Degree Busy

If the First-Second Degree Busy Indication is equipped, and the attendant attempts to Camp-on a call to a station in the ringing or dialing state, the attendant receives first degree busy indication. If the attendant attempts to Camp-on a call to a station that is second degree busy, Camp-on is not allowed. The attendant receives second degree busy indication.

Slow Answer Recall Modification

Slow Answer Recall Modification (SLAM) has an interaction after the attendant answers the recall. If SLAM is configured, then the target set is disconnected after the attendant answers the recall. The call is no longer camped on.

Feature packaging

Camp-On to a Set in Ringback or Dialing requires Camp-on French Type Approval (FRTA) package 197.

For network routing, the Network Attendant Service (NAS) package 159 is required.

If periodic Camp-on tone is desired, the Flexible Tones and Cadences (FTC) package 125 is required.

If music to the camped-on station is desired, the Music (MUS) package 44 is required.

For a station to Camp-on a trunk, the Station Camp-on (SCMP) package 121 is required.

Feature implementation

No change to existing configuration is required for the Camp-On to a Set in Ringback or Dialing feature.

Feature operation

See the Camp-On feature description contained within this document.

Camp-On to Multiple Appearance Directory Number

Contents

This section contains information on the following topics:

Feature description 907
Operating parameters 908
Feature interactions 908
Feature packaging 909
Feature implementation 909
Feature operation 909

Feature description

The Camp-On to Multiple Appearance DN enhancement allows a call camped on to an Multiple Call Ringing (MCR) or Multiple Call Non-ringing (MCN) Directory Number (DN) to be camped on to all sets with that DN. That is, any set with that MCR or MCN DN can receive the call when it idles. The camped call will Camp-On to each set as allowed for by the existing Camp-On feature. Also, each set with the MCR or MCN DN will receive Camp-On tone as long the camped call is in the Camp-On Queue. Prior to the introduction of the Camp-On to Multiple Appearance DN enhancement Camp-On was applied to the first set in the TN list.

This enhancement applies to Station Camp-On and Network Camp-On (regardless of where in the network the Camp-On originated).

An example of the sequence for multiple Camp-Ons to a single DN follows:

1 Sets A, B, and C are Meridian 1 proprietary telephones with the same MCR or MCN DN. All three sets are busy.

2 The attendant extends an external call to the busy DN and releases. Sets A, B, and C hear Camp-On tone.

3 A goes on-hook and camped call is presented to set A. Camp-On tone is removed from B and C.

4 If B went on-hook in step 3, the call is presented to set B and Camp-On tone is removed from sets A and C.

5 Similar operations as in step 4 if set C goes on-hook.

Operating parameters

The same feature requirements apply as for the Camp-On feature.

This Camp-On enhancement applies to Multiple-appearance Multiple-call Ringing (MCR) or Non-ringing (MCN) DNs; it does not apply to Multiple-appearance Single-call Ringing (SCR) or Non-ringing (SCN) DNs.

Feature interactions

Attendant Break-in

Camp-On will not be allowed on a set involved in an Attendant Break-in.

Centralized Multiline

This feature allows analog (500/2500 type) telephones to appear as MCR DNs. This enhancement should apply to these sets.

Make Set Busy

Camp-On will not be allowed on a set with Make Set Busy active.

Network Camp-On

The Camp-On enhancement applies to all Camp-On attempts regardless of where the Camp-On was originated.

Operator Call Back

Camp-On is not allowed on a set waiting for an Operator Recall signal.

Single Call Ringing and Non-ringing

Multiple-appearance Single Call Ringing and Non-ringing (SCR and SCN) DNs are not affected by the Camp-On enhancement.

Station Camp-On

The Camp-On enhancement applies to Station Camp-On.

Feature packaging

Camp-On to Multiple Appearance Directory Number requires International Supplementary Features (SUPP) package 131.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

For operating procedures, see the “Camp-On” feature description on page 911.

Camp-On

Content

This section contains information on the following topics:

Feature description 911
Operating parameters 911
Feature interactions 912
Feature packaging 916
Feature implementation 916
Feature operation 918

Feature description

The Camp-On feature routes one additional external call to a busy Directory Number (DN). When the attendant extends a call to a busy DN, the external call is camped-on to the telephone. If the Class of Service allows a warning tone, the user hears a tone indicating that a call is camped on. If the user frees the line within a specified time, the camped-on call rings the telephone automatically. If not, the call returns to the attendant as a recall.

Camp-On Tone is allowed or denied on a per-customer basis. The time a camped-on call waits is defined in LD 15 from 0 to 510 seconds, in multiples of two seconds. The default is 30 seconds.

Operating parameters

Camp-On applies to attendant-extended calls only. If the attendant hears a busy tone, another call has already been camped on to the busy telephone.

Feature interactions

Attendant Blocking of Directory Number

Camp-on will be denied for a DN that is blocked due to the Attendant Blocking of DN feature.

Attendant Break-In

If the destination DN has a camped-on incoming trunk call, the attendant cannot extend the urgent incoming call as a Camp-On call.

Call Forward All Calls
Call Forward, Internal Call

Call Forward All Calls and Internal Call Forward take precedence over Camp-On.

Call Forward/Hunt Override Via Flexible Feature Code

When a busy set is encountered, it is possible to Camp-on to the set, even if Call Forward/Hunt Override Via Flexible Feature Code has been activated

Call Forward No Answer

When the Call Forward No Answer timer expires for a ringing camped-on call, the call is given Attendant Recall treatment instead of Call Forward No Answer treatment.

Call Park Network Wide

When an attendant attempts to extend a call to a busy station across the network and the busy station returns a Camp-On allow signal, an attendant has the option of camping on a call or continuing with Network Call Park.

Call Park on Unsupervised Trunks

A Disconnect Timer applies to camped-on calls on all trunks on the route. All answered calls in the camped-on state will be disconnected if left in that state for an extended period.

Call Waiting
Call Waiting Redirection

If a Call Waiting Class of Service or key is defined, Camp-On cannot be provided.

Camp-On, Station

With Station Camp-On, any internal station can camp an external call on to another internal station that is busy. Prior to the introduction of this feature, attendants were the only parties that could camp calls on to busy internal stations. The term internal station includes stations on other nodes within a Meridian Customer Defined Network (MCDN). For more information, see the Camp-On, Station feature description.

China Number 1 Signaling - Called Party Control

A local attendant cannot Camp-on a call to an analog (500/2500 type) set that is on an outgoing trunk call that follows Called Party Control.

Enhanced Override
Forced Camp-On
Priority Override

Station-to-Station Camp-On and Attendant Camp-On are not affected by Forced Camp-On or Priority Override. The new Classes of Service (Camp-On From Another Telephone Allowed [CPFA], Camp-On From Another Telephone Denied [CPFD], Camp-On To Another Telephone Allowed [CPTA], and Camp-On To Another Telephone Denied [CPTD]) affect only Forced Camp-On. The Station Camp-On (SCMP) package (121) is required to return busy tone instead of ringback tone to the party camping on.

Flexible Feature Code Boss Secretarial Filtering

When an attendant is attempting to Camp-on a call to a boss set with filtering active, the call is routed to the secretary set, if the filtering is active for all calls. If filtering is active for external calls only, the call is routed to the secretary set if the call is an external call.

Flexible Voice/Data Terminal Number

Camp-On is not supported on data calls to a dynamic voice/ data TN.

Camp On is supported for voice calls to dynamic voice/data TN. However, no tone is inserted during a Camp On attempt if the Terminal Number is in a busy data mode.

Generic XFCOT Software Support

The Camp-On feature allows an attendant to route one additional call to a busy DN so it can be rung when it becomes free. If the busy DN is not free after a customer-defined time, the call is recalled to the attendant.

A call from a loopstart disconnect supervised or unsupervised loopstart trunk can be camped on. If a caller on an unsupervised loopstart trunk disconnects while the call is camped on, it is detected when the call is recalled or answered.

Caller disconnection during Camp-On operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the camped on call is dropped.

Group Hunt

Camping an incoming call on to a Pilot DN is not be supported

Group Hunting Queuing Limitation

No Camp-on tone is provided for Group Hunting Queuing Limitation.

Hunting

Hunting takes precedence over Camp-On.

Idle Extension Notification

When an extension is blocked for receiving calls due to the Idle Extension Notification feature, Camp-on is not possible.

Multi-Party Operations

Camp-on to a controlling party DN which is involved in a Consultation connection is not permitted. However, Camp-on is allowed at non-controlling parties DN's which are involved in the Consultation connection.

Multi-Party Operations Enhancements

Camp-on is allowed on a party receiving Patience Tone. Camp-on tone and is not applied to the party during Patience tone. However, Camp-on tone and is applied when the speechpath has been reestablished

Multi-Party Operations – Three-Party Service

While Camp-On is allowed to the party receiving the patience tone, the Camp-On tone is not applied to the party during the patience tone. The Camp-On tone is applied, however, when the speech path has been reestablished.

On Hold on Loudspeaker

Camp-On can be applied to a busy loudspeaker DN.

Override - Forced Camp-On and Priority Override Override, Enhanced

Station-to-Station Camp-On and Attendant Camp-On are not affected by Forced Camp-On or Priority Override. The new Classes of Service (Camp-On From Another Telephone Allowed [CPFA], Camp-On From Another Telephone Denied [CPFD], Camp-On To Another Telephone Allowed [CPTA], and Camp-On To Another Telephone Denied [CPTD]) affect only Forced Camp-On. The Station Camp-On (SCMP) package (121) is required to return busy tone instead of ringback tone to the party camping on.

Periodic Pulse Metering

Metered calls camped-on to a busy station by an attendant are charged against the attendant until the call is answered and the attendant releases.

Source Included when Attendant Dials

The source remains included while the attendant dials the destination.

Uninterrupted Line Connections

Warning Tone

Class of Service with warning tone denied allows a call to be camped on, but with no warning tone.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Camp-On tone for a customer.

2 LD 10 – Allow warning tone Class of Service for analog (500/2500 type) telephones.

3 LD 11 – Allow warning tone Class of Service for Meridian 1 proprietary telephones.

LD 15 – Enable Camp-On tone for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options data block.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	CTA	Enable Camp-On tone for the customer.
TYPE	TIM	Configure timers data block.
- RTIM	xx yy zz	Set recall timers. yy = Camp-On recall timer, response is 0-(30)-510.
...		
TYPE	FTR	Features and options
- STCB	(NO) YES	Station Camp-On Busy allowed.

- NSCP	(NO) YES	Network Station Camp-On to sets on this node allowed.

LD 10 – Allow warning tone Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	WTA	Allow warning tone.

LD 11 – Allow warning tone Class of Service for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	WTA	Allow warning tone.

Feature operation

To camp on an external call to a busy destination:

- Press **Rls**.
  The call is camped on to the extension. If you hear a busy tone, a call is already camped on the extension.

  If the call is not answered within a specified time, it recalls to the attendant. Both the Source and Destination indicators flash until the recall is answered. The call can be camped on again or released.

To answer a camped-on call, follows these steps:

1 When you hear a short beep indicating a camped-on call, hang up or press **Rls**.

2 When the telephone rings, lift the handset.
You are connected to the camped-on call.

Camp-On, Forced

Content

This section contains information on the following topics:

Feature description 919
Operating parameters 920
Feature interactions 922
Feature packaging 924
Feature implementation 924
Feature operation 927

Feature description

Forced Camp-On differs from normal Camp-On in that both internal and external calls can be camped on, rather than just external calls as with the Camp-On feature. The Forced Camp-On can be automatic or manual. The manual operation requires the use of the Enhanced Override (EOVR) key or Flexible Feature Code (FFC).

Forced Camp-On can be used as a feature by itself or in conjunction with Priority Override. The combination of the two features is referred to as Enhanced Override (EOVR).

For manual Forced Camp-On, an analog (500/2500 type) telephone user has to dial the EOVR Flexible Feature Code (FFC), while a Meridian 1 proprietary telephone user has to use the EOVR key.

A second operation of the EOVR key or FFC executes Enhanced Override.

Forced Camp-On is similar to station-to-station Camp-On except that Forced Camp-On can be done with either no call on hold or an external or internal call on hold. It can be done automatically or manually; whether it is automatic or manual is determined by the response to the Automatic Forced Camp-On (AFCO) prompt in LD 15.

For manual operation, once a busy telephone has been reached, the first depression of the EOVR key or the first dialing of the EOVR FFC attempts Forced Camp-On. If successful, Forced Camp-On introduces Camp-On tone into the connection. If unsuccessful, overflow (fast busy) tone is returned to the party attempting the Forced Camp-On.

For Forced Camp-On to be attempted, all other methods of call termination must have been tried, the last of which was Camp-On. If station-to-station Camp-On or automatic Forced Camp-On has occurred, or Forced Camp-On has been excluded by the new telephone options, then the first depression of the EOVR key or dialing of the EOVR FFC executes Enhanced Override. If, however, Forced Camp-On is denied due to existing Camp-On restrictions, Enhanced Override is also denied.

Operating parameters

The Flexible Feature Codes (FFC) package (139) must be equipped for Forced Camp-On and Priority Override to be available from analog (500/2500 type) telephones.

For analog (500/2500 type) telephone activation, the Multi-Party Operations (MPO) package (141) must be equipped, with "YES" as the response to the RALL prompt in LD 15 to ensure register recalls are required before dialing control digits. The EOVF FFC defined must not start with the same digit as one of the control digits. The control digits are defined in LD 15 and are printed as part of the Customer Data Block (LD 21).

Telephones or trunks involved in any of the following cannot be camped on to:

- Non established call
- Conference call
- Attendant call

- Attendant call using Centralized Attendant Service (CAS), Primary Rate Interface (PRI), or Integrated Services Digital Network (ISDN) trunk
- Make Set Busy
- Do Not Disturb
- Automatic Call Distribution (ACD) call
- Operator Call Back
- Hold
- Data call
- Release Link call, and
- Parked call.

Call Forward and Hunting take precedence over Call Waiting. If Call Waiting is allowed, Camp-On is not attempted. If Call Waiting is not allowed, station-to-station Camp-On is automatically attempted. If this succeeds, Enhanced Override can still follow. If Camp-On fails because there is no external call, Forced Camp-On and Enhanced Override may still work. However, if Camp-On fails due to other limitations, Forced Camp-On and Enhanced Override will also not work.

Even though Camp-On will still function when Warning Tone Denied (WTD) Class of Service is defined, Forced Camp-On requires Warning Tone Allowed (WTA) Class of Service.

Camp-On requires an external call on hold. Forced Camp-On can be done without a call on hold, or with both internal and external calls on hold.

Camp-On Classes of Service (Camp-On From another telephone Allowed [CPFA], Camp-On From another telephone Denied [CPFD], Camp-On To another telephone Allowed [CPTA], and Camp-On To another telephone Denied [CPTD]) apply to Forced Camp-On and automatic Forced Camp-On (AFCO) only. They do not apply to Station or attendant Camp-On.

If a telephone is denied Forced Camp-On by Class of Service, Enhanced Override may still be attempted.

Feature interactions

Attendant Break-In

Telephones with a toll operator break-in call cannot be camped on to. Overflow tone is returned to telephones attempting Forced Camp-on.

Attendant Calls

Telephones involved in attendant calls cannot be camped on to. Overflow (fast busy) tone is returned to telephones on which Forced Camp-On is attempted.

Automatic Call Distribution

Telephones involved in Automatic Call Distribution calls cannot be camped on to. Overflow (fast busy) tone is returned to telephones attempting Forced Camp-On.

Call Hold, Deluxe
Call Hold, Permanent
Hold

Neither held calls nor telephones with calls on hold can be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On.

Camp-On

Station-to-Station Camp-On and attendant Camp-On are not affected by Forced Camp-On. The Classes of Service (Camp-On From another telephone Allowed [CPFA], Camp-On From another telephone Denied [CPFD], Camp-On To another telephone Allowed [CPTA], and Camp-On To another telephone Denied [CPTD]) affect only Forced Camp-On. The Station Camp-On (SCMP) package (121) is required to return busy tone instead of ringback tone to the party camping on.

Conference calls

Telephones involved in Conference calls cannot be camped on to. Overflow tone is returned to telephones attempting Forced Camp-On.

Data calls

Data calls have Warning Tone Denied (WTD) Class of Service and therefore cannot be camped on to. Overflow tone is returned to telephones attempting Forced Camp-On.

Do Not Disturb

Telephones with Do Not Disturb enabled cannot be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On.

Make Set Busy

Telephones with Make Set Busy active cannot be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On. Voice Call is blocked by Make Set Busy.

Multi-Party Operations

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

Night Restriction Classes of Service

If Forced Camp-on and Night Restriction for Forced Camp-on Class of Service (NRCA) are assigned, Forced Camp-on will be operational for the set only when Night Service is in effect.

Operator Call Back

Telephones involved in an Operator Call Back call or Toll Operator Break-In cannot be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On.

Override

When Priority Override is activated, it replaces normal override. Once Priority Override has been performed on a set, its Digit Display shows the DN of the overriding set.

Feature packaging

Forced Camp-On requires the following packages to function as described in this document:

- Station Camp-On (SCMP) package 121
- Flexible Feature Codes (FFC) package 139
- Priority Override/Forced Camp-On (POVR) package 186

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the customer for Automatic Forced Camp-On and Station Camp-On tone.

2 LD 57 – Configure Enhanced Override Flexible Feature Codes (FFC).

3 LD 10 – Configure analog (500/2500 type) telephones for Forced Camp-On.

4 LD 11 – Configure Meridian 1 proprietary telephones for Forced Camp-On.

5 LD 14 – Configure trunks for Forced Camp-On.

LD 15 – Configure the customer for Automatic Forced Camp-On and Station Camp-On tone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	MPO	Multi-Party Options
...		

- AFCO	(NO) YES	Automatic Forced Camp-On. Enter YES if Forced Camp-On is to be applied automatically. Enter NO if Forced Camp-On is to be applied manually.
...		
TYPE:	FTR	Features and options
- STCB	(NO) YES	Station Camp-On Busy tone. Enter NO if Busy Tone is not to be given to the transferring (controlling) party when the desired station is busy. Enter YES if Busy Tone is to be given to the transferring (controlling) party when the desired station is busy.

LD 57 – Configure Enhanced Override Flexible Feature Codes (FFC).

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FFC	Flexible Feature Codes.
...		
CODE	EOVR	Enhanced Override (programmable only when the Priority Override package 186 is equipped).
EOVR	y...y	y...y is a one- to seven-character input that the user must dial to use the FFC. Valid inputs are digits 0 through 9, asterisk (*), and octothorpe (#).

LD 10 – Configure analog (500/2500 type) telephones for Forced Camp-On.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Telephone type.

CLS	(CPFA) CPFD	Forced Camp-On from another telephone to this telephone (Allowed) Denied.
	(CPTA) CPTD	Forced Camp-On to another telephone from this telephone (Allowed) Denied.
	WTA	Warning Tone Allowed.

LD 11 – Configure Meridian 1 proprietary telephones for Forced Camp-On.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN		Terminal Number
	l s c u	For Large Systems
	c u	For Small Systems and Succession 1000 systems
CLS	(CPFA) CPFD	Forced Camp-On from another telephone to this telephone (Allowed) Denied.
	(CPTA) CPTD	Forced Camp-On to another telephone from this telephone (Allowed) Denied.
	WTA	Warning Tone Allowed.
...		
KEY	xx EOVR	Add an Enhanced Override key, where; xx = the key number (allowed to be programmed only if Priority Override package 186 is equipped).

LD 14 – Configure trunks for Forced Camp-On.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
...		
CLS	WTA	Warning Tone Allowed.

Feature operation

Forced Camp-On can be used when making either a simple or consultation call (that is, having a call on hold while calling another party). The following feature operation descriptions use telephone A (an analog (500/2500 type) telephone) or telephone E (a Meridian 1 proprietary telephone) to call telephone B, which is connected to party C.

The telephones are configured as follows:

1 Telephone A is an analog (500/2500 type) telephone with Warning Tone Allowed (WTA) Class of Service.

2 Telephone B has Warning Tone Allowed (WTA) Class of Service.

3 Party C has Warning Tone Allowed (WTA) Class of Service and can be any telephone type or a Direct Inward Dial (DID), TIE, or Central Office (Public Exchange) (COT) trunk.

4 Telephone E is a Meridian 1 proprietary telephone with Warning Tone Allowed (WTA) Class of Service and an Enhanced Override (EOVR) key equipped.

For the following examples:

1 Telephones A and E both have Camp-On From another telephone Allowed (CPFA) Class of Service.

2 Both telephone B and telephone C are involved in a simple call, not a conference call.

3 Telephone B has Camp-On To another telephone Allowed (CPTA) Class of Service.

4 Call Forward, Hunting, and Call Waiting are not in use.

In the following feature operation descriptions, the term “recall” refers to performing a register recall, which can be performed in a number of different ways. Some typical examples are:

- Flash the switchhook. (This is the equivalent of hanging up the handset and picking it back up. This on hook, off hook is performed in a time period that is less than what the system would consider to be a valid disconnect.)
- Press the flash or LINK button if equipped.

The Camp-On tone is always provided for Forced Camp-On since Warning Tone Allowed (WTA) Class of Service is a prerequisite. This tone can be a buzz for Meridian 1 proprietary telephones or a single burst of tone for analog (500/2500 type) telephones if the customer (LD 15) option Periodic Camp-On Tone Denied (CTD) is selected. If the customer (LD 15) option Periodic Camp-On Tone Allowed (CTA) is selected, the Camp-On Tone as defined in the Flexible Tones and Cadences (FTC) (LD 56) in response to the CAMP prompt will be used.

While camping on, the party attempting the Camp-On, either telephone A or E, receives ringback if the Station Camp-On (SCMP) package (121) is not equipped, or either ringback or busy tone, as defined by the response to the Station Camp-On Busy tone (STCB) prompt in LD 15, if the SCMP package is equipped.

Forced Camp-On with an analog (500/2500 type) telephone

With automatic Forced Camp-On turned off; response to AFCO in LD 15 was "NO":

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A receives busy tone.
3	A performs a recall.	A receives special dial tone (SDT).
4	A dials EOVR FFC to attempt Forced Camp-On.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise B receives Camp-On tone and A receives ringback or busy tone depending on the options equipped. A is manually Forced Camp-On to B.
5	B disconnects from the call.	Telephone A rings telephone B.

With automatic Forced Camp-On turned on; response to AFCO in LD 15 was "YES":

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A attempts Forced Camp-On to B.
3	If Forced Camp-On was successful.	A receives ringback or busy tone depending on the options equipped. A is automatically Forced Camp-On to B.
4	B disconnects.	A rings B.

Forced Camp-On with a Meridian 1 proprietary telephone

With automatic Forced Camp-On turned off; response to AFCO in LD 15 was "NO":

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E receives busy tone.
3	E presses EOVR key to attempt Forced Camp-On.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise B receives Camp-On tone and E receives ringback or busy tone depending on the options equipped. E is manually Forced Camp-On to B.
4	B disconnects from the call.	Telephone E rings telephone B.

With automatic Forced Camp-On turned on; response to AFCO in LD 15 was "YES":

	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E attempts Forced Camp-On to B.
3	If Forced Camp-On was successful.	E receives ringback or busy tone depending on the options equipped. E is automatically Forced Camp-On to B.
4	B disconnects.	E rings B.

Camp-On, Station

Contents

This section contains information on the following topics:

Feature description 931
Operating parameters 932
Feature interactions 933
Feature packaging 933
Feature implementation 934
Feature operation 934

Feature description

With this feature, any internal station can camp an external call on to another internal station that is busy. Prior to the introduction of this feature attendants were the only parties that could camp calls on to busy internal stations. The term internal station includes stations on other nodes within a Meridian Customer Defined Network (MCDN).

When a transferring party reaches a busy internal party, the transferring telephone will receive Ringback tone (providing certain conditions are met). When the transferring party completes the transfer, the external (calling) party will Camp-On to the desired party and the external party (an external party is any CO, DID, FEX, or WATS call) will receive ringback tone or music.

This feature applies to both standalone and network environments.

Within a network environment, the transferring and camped on to stations may be on the same or different nodes, as long as all nodes are configured with Network Station Camp-On.

Operating parameters

The restrictions that currently apply to the operation of the Camp-On feature from an Attendant Console also apply to Station Camp-On.

These restrictions are:

- Camp-On is not permitted if the desired station is in a state other than established (that is, ringing, dialing).
- Only one call at a time can Camp-On to a busy station.
- Calls cannot Camp-On to a station with the Call Waiting feature configured.
- The station camped on to will be given Warning tone only if the customer has Camp-On Tone Allowed (CTA) in the Customer Data Block (LD 15) and the station has Warning Tone Allowed (WTA) Class of Service assigned. If the station has Warning Tone Denied (WTD) Class of Service assigned, the Camp-On will take effect without giving any Camp-On tone to the camped on to (desired) party.
- The transferring station will receive Busy tone only if the response to the STCB prompt in the Customer Data Block (LD 15) of the camped on to (desired) telephone is YES. Otherwise, the transferring station will receive Ringback tone.

Camp-On indication

When a call is extended from an attendant to a busy station there is a specific combination of tones and indicator states to identify the Camp-On state.

When an inquiry call is made from a station, there is only one way for the user to distinguish between a busy telephone and an idle ringing telephone. That way is to ensure that the response to the STCB prompt in the Customer Data Block (LD 15) of the camped on to (desired) telephone is YES. Otherwise, Ringback tone is provided in both cases.

Feature interactions

Call Forward All Calls
Call Forward Busy
Call Waiting
Hunting

Call Waiting, Call Forward Busy (for DID calls only), Call Forward All Calls, Call Waiting and Hunting all take precedence over Station Camp-On.

Camp-On

With Station Camp-On, any internal station can camp an external call on to another internal station that is busy. Prior to the introduction of this feature, attendants were the only parties that could camp calls on to busy internal stations. The term internal station includes stations on other nodes within a Meridian Customer Defined Network (MCDN). For more information, see the Camp-On, Station feature description.

Dial Impulse Set

A Dial Impulse (DIP Class of Service) station must have TSA Class of Service to perform a Station Camp-On.

Network Attendant Service

For network-wide Station Camp-On, NAS must be equipped at each node of the network.

Feature packaging

For standalone environments, the Station Camp-On (SCMP) package 121 is required.

For network environments, the Station Camp-On (SCMP) package 121 and the Network Attendant Service (NAS) package 159 are required.

For Music (MUS), package 44 is required.

Feature implementation

LD 15 – Configure Station Camp-On for both standalone and network environments.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	FTR	Features and options
...		
- STCB	(NO) YES	Station Camp-On Busy tone. Enter NO if Busy tone is not to be given to the transferring (controlling) party when the desired station is busy. Enter YES if Busy tone is to be given to the transferring (controlling) party when the desired station is busy.
- NSCP	(NO) YES	Network Station Camp-On. Enter NO if telephones on this node are not allowed to have calls camped on by telephones in other nodes. Enter YES if telephones on this node are allowed to have calls camped on by telephones in other nodes.

Feature operation

Standalone case

Any station, not necessarily the Night DN, attempting to transfer an external call may, during the associated inquiry call, camp the trunk on to a busy station.

The Camp-On will take effect from the moment the transferring station has completed the transfer to the desired DN.

The transferring station will hear Ringback tone or Busy tone depending on the option entered in response to the STCB prompt in the Customer Data Block (LD 15). This prompt applies to any telephone, not just the Night DN. By default (STCB is set to NO), the transferring party will hear Ringback tone.

The desired station will hear Camp-On tone if it has WTA Class of Service assigned. Otherwise, if it has WTD Class of Service, the Camp-On will take effect without the desired party being informed a call is camped on.

When the transfer is completed, the external party is camped on to the desired station and receives either Ringback tone or an announcement.

Network case

Any station, not necessarily the Night DN, attempting to transfer an external call across an MCDN network may, during the associated inquiry call, Camp-On the trunk to a busy station.

The location of the transferring party has no effect on the Station Camp-On capability.

The Camp-On will take effect from the moment the transferring station has completed the transfer to the desired DN.

The transferring station will hear Ringback tone or Busy tone depending on the option entered in response to the STCB prompt in the Customer Data Block (LD 15). This prompt applies to any telephone, not just the Night DN. By default (STCB is set to NO), the transferring party will hear Ringback tone. The tone given, either ringback tone or Busy tone, is determined by the node in which the desired (camped on to) party resides.

The desired station will hear Camp-On tone if it has WTA Class of Service assigned. If it has WTD Class of Service, the Camp-On will take effect without the desired party being informed a call is camped on.

When the transfer is completed, the external party is camped on to the desired station and receives either Ringback tone or an announcement.

Recall timing on Camp-On calls

When any station extends an external call, recall timing will be initiated if the call is camped on to a busy station.

The recall timing will start from the moment that the extending station “releases” the call. The value of the recall timer is set by the prompt RTIM in the Customer Data Block (LD 15).

At the recall, the camped on call will be routed to the attendant. If the attendant is in Night Service, Night treatment is given; if NAS routing is active, the call will be routed according to the NAS configuration.

Standalone case

When the recall to the attendant occurs, the Camp-On is canceled. If the attendant is busy during the recall, the recall will be queued.

Network case

When the recall occurs and the attendant has answered the recall, the call will still be camped on to the desired party. If during the recall the attendant is busy, the recall will be queued.

Card LED Status

Contents

This section contains information on the following topics:

Feature description 937
Operating parameters 938
Feature interactions 938
Feature packaging 938
Feature implementation 938
Feature operation 939

Feature description

This feature allows the use of Swedish Televerket (TVT) peripheral equipment on the system. This is accomplished by defining individual terminal loops as TVT type in LD 17. The system software is modified to allow the status (on/off) of the LED on the faceplate of the TVT cards to be opposite of the LED on NT cards. The TVT peripheral cards (standard extension line, off-premises extension and Multi Frequency Receiver (MFR)) are equivalent to the NT 500-type line card and Digitone Receiver (DTR). Since the TVT off-premises extension line card must be treated as a local extension by the system, the OPX Class of Service is prohibited for this card in LD 10.

Operating parameters

The system software supports the following TVT cards:

- single-density standard extension line card (TPC60)
- dual-density extension line card (TPC451)
- two-wire Off-premise Extension (OPX) line card (TPC22)
- four-wire Off-premise Extension (OPX) line card (TPC23), and
- multi-frequency receiver (MFR) card (TPC15).

Feature interactions

Card LED Status does not interact with other features.

Feature packaging

Card LED Status requires International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure the system hardware and software parameters.

2 LD 10 – Create or modify data blocks for analog (500/2500 type) telephones.

LD 17 – Configure the system hardware and software parameters.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	CEQU	Common Equipment parameters
...		
CEQU	(NO) YES	Change CE parameters.
TERM	T0-T159	TVT single density local terminal loops.

LD 10 – Create or modify data blocks for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Telephone type.
...		
CLS	OPN	Allows Swedish TVT off-premise line card to be treated as a local extension by the system.

Feature operation

No specific operating procedures are required to use this feature.

Centralized Multiple Line Emulation

Contents

This section contains information on the following topics:

Feature description 941
Operating parameters 942
Feature interactions 942
Feature packaging 942
Feature implementation 943
Feature operation 944

Feature description

The Centralized Multiple Line Emulation (CML) feature allows a mixed group of telephones (analog (500/2500 type) telephones, or Meridian 1 proprietary telephones) to answer calls ringing at a central DN (referred to as the CML DN). This function is provided by using the Automatic Call Distribution (ACD) and Multiple Appearance Directory Number (MADN) features, and making modifications to the Call Pick-up feature.

Large queues to the CML DN (up to 15 calls) are handled by the ACD feature, which distributes the calls to members of the CML group.

Small queues to the CML DN (one or two calls) are handled using MADNs configured on a Meridian 1 proprietary telephone.

Operating parameters

The Centralized Multiple Line Emulation feature is not supported by Attendant Administration.

Call Pick-up groups assigned at the key level cannot be given a group number containing ACD DNs, since calls ringing in an ACD queue cannot be picked up.

Calls cannot be picked up from a station having direct-termination-denied Class of Service.

Normal tenant-service calling restrictions apply. If a station cannot receive a ringing call, then the call cannot be picked up for that station. A station that cannot direct dial another station cannot pick up a call from that station.

Calls ringing on the CML priority station are picked up before ringing Central Office trunk calls in the same Call Pick-up group.

Feature interactions

Digit Display

The digit display of the station picking up a parked call recall shows the parked call's access code followed by the parked call's access-identification code. If the picked-up call is a group member call, the display shows the group number of the picked-up station.

Feature packaging

Centralized Multiple Line Emulation requires International Supplementary Features (SUPP) package 131.

The following packages are also required:

- Basic Automatic Call Distribution (BACD) package 40
- Network Priority Queuing (PQUE) package 60

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Configure the Required Classes of Service.

2 LD 11 – Assign ringing number pickup groups to keys.

LD 10 – Configure the Required Classes of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Telephone type.
...		
CLS	(PRSD) PRSA	Priority Call Pick-up station (denied) allowed.
	(CRD) CRA	Continuous Ringing (denied) allowed.
	(MCRD) MCRA	Multiple Call Arrangement (denied) allowed.

LD 11 – Assign ringing number pickup groups to keys.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
...		
RNPG	(0)-4095	Ringing Number Pick-up Group. Respond with the number of the Ringing Number Pick-up group for which the set is to be assigned. To remove a telephone from a group, enter 0 in response to the RNPG prompt.

...		
CLS	(PRSD) PRSA	Priority Call Pick-up station (denied) allowed.
...		
KEY	xx RNP yyyy	xx = Key number RNP = Ringing Number Pick-up yyyy = Ringing Number Pick-up group number (optional). If the group number is not entered, the key will pick-up the group number assigned to the station. If the group number is entered, the key will pick-up calls in the specified group yyyy.

Feature operation

The ability to notify a large group that a CML is ringing is provided through modification of the Call Pick-up feature. A ringer, centrally mounted on a wall, rings whenever a call comes into the CML DN, and continues to ring until the call is answered. The ringer is configured as a priority 500-type set, which ensures that a call ringing on the CML DN is answered before any other station ringing in the Call Pick-up group.

1 To answer a CML call using a Meridian 1 proprietary telephone, press the **RNP** key.

To answer a call in your Call Pickup group from an analog (500/2500 type) telephone, follow these steps:

1 Lift the handset.

2 Dial the PURN FCC.

Centrex Switchhook Flash

Contents

This section contains information on the following topics:

Feature description 945
Operating parameters 946
Feature interactions 947
Feature packaging 948
Feature implementation 949
Feature operation 952

Feature description

Centrex Switchhook Flash (THF) permits the user to signal the Central Office (CO)/Public Exchange during an established CO call, requesting activation of a Central Office based service. Such services can include Call Transfer, Three-way Calling, Malicious Call Trace, Conference, or Autodial Tandem Transfer. For more information on these services, please refer to the feature descriptions contained within this document.

The feature is useful when Centrex is the backbone of the service network. Centrex Switchhook Flash (THF) is supported by the following trunk types:

- Automatic Identification of Outward Dialing (AIOD)
- Common Control Switching Arrangement, Automatic Number Identification (CCSA ANI)
- Centralized Automatic Message Accounting (CAMA)

- Central Office (CO)
- Common Control Switching Arrangement (CCSA)
- Direct Inward Dial (DID)
- Foreign Exchange (FX)
- Wide Area Telephone Service (WATS)
- CO trunks located at a remote node connected via ISDN Meridian Customer Defined Network (MCDN) TIE trunks, and
- Analog, Digital Trunk Interface (DTI), and DT12 CO trunks.

Whenever Centrex Switchhook Flash (THF) is invoked, the system checks for the following:

- With analog (500/2500 type) telephones, that the Class of Service supports THF. With Meridian 1 proprietary telephones, the feature cannot be activated if a corresponding key is not equipped.
- That the telephone is on an active two-way trunk call.
- That THF is enabled in the trunk's Class of Service.

If any of the above checks fails, the user hears an overflow tone. After the tone times out, the original connection resumes.

Operating parameters

This feature is not supported on Basic Rate Interface (BRI) telephones.

THF is not supported on Primary Rate Interface (PRI) or BRI Central Office trunks.

On Meridian 1 proprietary telephones, once the THF key has been pressed, all other function keys are blocked. While waiting for the Centrex connection, only the RLS key or on-hook connection is operative. Pressing the RLS key or hanging up terminates the original connection as well as the THF message.

For the analog (500/2500 type) telephones, another switchhook flash is not allowed once THF has been invoked. A second switchhook flash is treated as an on hook disconnection.

Only voice calls are supported on THF.

In Italy the DTI2 register recall signal is currently only supported for Type Approval and is not commercially available.

The 1.5 or 2 Mbit Digital Trunks Interface pack is required.

Because the software cannot recognize which type of Intelligent Peripheral Equipment (IPE) CO line card (for example, XCOT, or XFCOT) is in use, CO trunks belonging to different card types should not coexist on the same Route Data Block (RDB).

- The minimum value of the range for the Flash-length (FLH) timer for a Centrex Switchhook Flash, defined in LD 16 in response to the TIMR prompt, is 60-1536 milliseconds.
- Attendant Consoles can activate the feature.
- The THF feature can be activated on DTI2 Central Office trunks and Intelligent Peripheral Equipment (IPE) Central Office trunks.
- The Centrex Switchhook timing on the Extended Flexible Universal Trunk (EXUT) is performed using firmware, offering a significant improvement in trunk timing accuracy.

Feature interactions

Autodial Tandem Transfer

Because Autodial Tandem Transfer uses Centrex Switchhook Flash (THF), it is affected by any modification to the THF enhancement feature.

China – Attendant Monitor

If any set at the customer location involved in the monitored call switchhook flashes or performs a Centrex switchhook flash, Attendant Monitor is immediately deactivated.

Collect Call Blocking

A Centrex Switchhook Flash cannot be invoked by another feature while the Collect Call Blocking answer signal is being sent.

Conference

THF allows conference calls through the CO. It can be invoked only if there is an established call connected to an outside trunk. If the telephone is engaged in internal conference calls, THF cannot be used.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

If an analog (500/2500 type) telephone is part of an Executive Intrusion conference, any Switchhook Flash is ignored.

Malicious Call Trace - Enhanced

Interaction with the Centrex switchhook flash results because the flash range is changed for this feature. Communication to the CO (trunk hook flash) is performed by using the Centrex switchhook flash feature base code. The enhanced range is available for the Centrex switchhook flash.

Periodic Clearing on RAN, Meridian Mail, ACD and Music

This feature enhancement is not supported if used together with Centrex Switchhook flash.

Secrecy

If secrecy is not allowed in LD 15 (OPT = SYD), the attendant must use the EXCL DEST or EXCL SRC keys to select the Central Office trunk on which the THF has to be sent. The THF is not activated when both SRC and DEST are included.

Feature packaging

This feature is included in base System Software. Centrex Switchhook Flash (THF) package 157 has no package dependencies. The End-to-End Signaling (EES) package 10 is recommended for users with Meridian 1 proprietary telephones, and Attendant Consoles.

NOTE: If both THF and the 2 Mbit Digital Trunk Interface (DTI2) package 129 are present, this feature can also be applied to digital Central Office trunk connections.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable Centrex Switchhook flash for analog (500/2500 type) telephones..

2 LD 11 – Enable Centrex Switchhook Flash for Meridian 1 proprietary telephones.

3 LD 12 – Enable Centrex Switchhook Flash for attendant consoles.

4 LD 14 – Enable Centrex Switchhook Flash for each trunk.

5 LD 16 – Enable Centrex Switchhook Flash for each trunk route.

6 LD 73 – Activate the THF on digital trunks for incoming and outgoing calls.

LD 10 – Enable Centrex Switchhook flash for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	THFA THFD	Allow Centrex Switchhook Flash. THFD = Default

LD 11 – Enable Centrex Switchhook Flash for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx THF	Add a Centrex Switchhook Flash key; xx is the key number.

LD 12 – Enable Centrex Switchhook Flash for attendant consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Console type.
...		
KEY	xx THF	Add a Centrex Switchhook Flash key; xx is the key number.

LD 14 – Enable Centrex Switchhook Flash for each trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AID	Automatic Identification of Outward Dialing (AIOD) trunk data block.
	CAA	Common Control Switching Arrangement Automatic Number Identification (CCSA ANI) trunk data block.
	CAM	Centralized Automatic Message Accounting (CAMA) trunk data block.

	COT	Central Office (CO) trunk data block.
	CSA	Common Control Switching Arrangement access line data block.
	DID	Direct Inward Dialing (DID) trunk data block.
	FEX	Foreign Exchange trunk data block.
	WAT	Wide Area Telephone Service trunk data block.
CLS	THFA THFD	Allow Centrex Switchhook Flash. THFD = Default

LD 16 – Enable Centrex Switchhook Flash for each trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CNTL	YES	Change controls or timers
- TIMR	FLH <space> 60-(510)-1536	Flash timer in msec. The range of the Centrex switchhook flash timer is 60-(510)-1536. The FLH value is rounded down to the nearest 10 msec. tick. If the value entered is 128 or 129, then it is set to 130 msec. *Software controlled flash* 60-127 msec. Digit 1 will be sent. 128-1536 msec. software controlled switchhook flash. ***Note:*** A FLH timer value of 127 msec. or less is not supported by the XFCOT card. The firmware controlled flash must be used. *Firmware controlled flash* The user can enter any value from 60 to 1536 msec. 90 msec. is the hardcoded firmware flash for an XFCOT card; the technician should enter 90 msec. ***Note:*** the FWTM prompt must be set to YES for the trunk associated with this route in LD 14, if firmware timing is to be used.

LD 73 – Activate the THF on digital trunks for incoming and outgoing calls.

Prompt	Response	Description
REQ	NEW CHG	New Change
TYPE	DTI2	2 Mbits DTI
FEAT	ABCD	Digital signaling category.
SICA	XX	Signaling category table.
...		
...		
P RRC(S)	ABCD	Register Recall signal.
TIME	10-(100)-630	Duration of RRC Pulse in msec.

Feature operation

Analog (500/2500 type) telephones

To use Centrex Switchhook Flash (THF) from an analog (500/2500 type) telephone, follow these steps:

1. Flash the switchhook to receive a special dial tone.
2. Enter the Special Prefix (SPRE) code, then the THF feature access code (96). Alternatively, the appropriate Flexible Feature Code (FFC) should be used.

To reestablish a connection before the overflow tone ends, flash the switchhook.

Meridian 1 proprietary telephones

To use Centrex Switchhook Flash (THF) from a Meridian 1 proprietary telephone, press the key configured for THF.

To reestablish a connection before the overflow tone ends, press the DN key or the key establishing the original call.

Attendant Consoles

Attendant Consoles must use the THF key. Dial access is not supported on these consoles.

To reestablish a connection before the overflow tone ends, press the DN key or the key establishing the original call.

Charge Account and Calling Party Number

Contents

This section contains information on the following topics:

Feature description 955
Operating parameters 957
Feature interactions 957
Feature packaging 959
Feature implementation 960
Feature operation 962

Feature description

Used in conjunction with Call Detail Recording (CDR), Charge Account bills calls directly to specific accounts or charge numbers instead of Directory Numbers (DN).

Charge Account supports fixed-length numbers of 0 to 23 digits (default is 0), specified on a per-customer basis. The charge account number is validated by the system for length only. Verification of the actual digits entered is part of CDR downstream processing.

On Meridian 1 proprietary telephones, this feature can be activated by a separate Charge key/lamp pair, or dial accessed. On Attendant Consoles, it is activated by a separate key/lamp pair. On single-line telephones, it is dial-accessed.

When a Charge Account number is used, the entire call is billed to that number. The number can be entered either before or during a call, or when Consultation Hold, Call Transfer, or Conference is activated.

The Charge Account feature is not supported for internal calls. A Charge Account number entered through the Charge key/lamp pair is blocked for established internal calls.

Charge Account can be used to charge an entire conference call or portions of the call. Portions of the call are assigned to different accounts by entering the account number when adding trunks to a conference, before the conference is completed.

- When using analog (500/2500 type) telephones, enter the account information immediately after the switchhook flash, before the new trunk is dialed.
- When using Meridian 1 proprietary telephones, enter the number after pressing the Conference key the first time, and before dialing.

The charge record shows the identity of the user who made the entry and the trunk that was added to the call. If the new call is not added to the conference, the record shows a simple two-party call.

An entire call is charged to the same account by entering the charge number while active on the conference. When using Meridian 1 proprietary telephones, press the Charge key and enter the number in the usual manner. When using analog (500/2500 type) telephones, enter the number after a switchhook flash.

The call is reestablished without dialing additional trunks; a record is produced for each trunk involved in the conference. In all these records, the telephone user entering the number is considered the originating party. When an entire call is charged to only one account number, it must be entered while all trunks are connected to the conference.

Calling Party Number (CPN) is an extension of Charge Account that allows entry of the calling party's number on collect calls. Meridian 1 proprietary telephones are assigned a separate Calling Party Number (CPN) key/lamp pair to activate this feature. When the calling party's number is entered, a Calling Party Number (CPN) record is produced. This record may be compared to a telephone company billing for collect calls. Calling party numbers can be up to 23 digits, and may include an asterisk (*) and octothorpe (#). A CPN record is generated on the Call Detail Recording (CDR) device similar to a normal Charge record.

Operating parameters

A valid charge account number is recognized when the number of dialed digits matches the account length, or when the octothorpe (#) indicates end of dialing. After a valid charge account number has been entered, the system returns a dial tone.

If too few digits are dialed, no response is given until the interdigit timeout occurs. Overflow tone is returned for 15 seconds after timeout; then the user is locked out.

If Call Transfer or Conference is used to consult with a third party and returns to the original call without completing the transfer or conference, the charge account number is applied to the Consultation call only.

Attendant use of Charge or CPN is restricted to situations in which there is only one account party involved in the call (source side). When the calling party number is used, the attendant must transfer the call, or the Call Detail Recording (CDR) record does not reflect it.

Feature interactions

Attendant Barge-In
Attendant Busy Verify

A charge account number cannot be entered when Barge-In or Busy Verify is active. Barge-In cannot be used to connect to a trunk after an account number has been entered.

Call Transfer

A Call Transfer call produces two records: a Call Detail Recording (CDR) start record and a CDR end record.

China – Flexible Feature Codes - Outgoing Call Barring

Digits dialed after a charge account are checked against the active Outgoing Call Barring level.

Conference

Conference calls produce multiple Call Detail Recording (CDR) records. Whenever a new trunk is added to a conference, the connection between the connected telephone and the trunk is recorded, and a connection to the conference loop is established. This causes CDR to generate a start record with the telephone and trunk identified as the involved parties. As trunks are removed from a conference, CDR end records are produced. These records may identify different telephones or conferences as the local parties.

Music, Enhanced

The Charge Account (CHG) and Calling Party Number (CPN) keys place the far end party on Hold while a charge number is entered. The held party receives Music during this period.

Override

When Charge Account is used during active Override, some digits may be lost. When entered with Override in conference, a Charge Account number is accepted and no digits are lost.

Ring Again

When Ring Again is activated, no charge record is generated, but the information is stored for future use. If Ring Again is canceled before a trunk is seized, the charge number is deleted and no record is produced. If a trunk is seized later by Ring Again, the charge record is generated in the usual manner. The use of Ring Again with Charge Account ties up system resources because an auxiliary call register must be maintained in the Ring Again queue.

Speed Call

Charge account numbers, including the Charge Account access Special Prefix (SPRE) code, can be stored as Speed Call or Autodial numbers. All current limitations of these features apply, such as a maximum of 23 digits per entry, including the access code. An Autodial number or dialed digits can follow, but not precede, a Speed Call number. The digits generated by an Autodial key during feature operation are accepted as Charge Account digits.

Telephone keys

A Charge Account entry is aborted with any of the following keys:

- DN
- Page
- Voice Call
- In-Calls
- Call Waiting
- Call Pickup
- Release
- Not Ready
- a loop key
- Release Destination, and
- Release Source.

Feature packaging

CDR with Charge Account (CHG) package 23 requires:

- Call Detail Recording (CDR) package 4
- Charge Account/Authorization Code Base (CAB) package 24

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Add or modify the customer Charge Account.

2 LD 10 – Allow analog (500/2500 type) telephone access to Charge Account.

3 LD 11 – Allow Meridian 1 proprietary telephone access to Charge Account.

4 LD 12 – Allow attendant console access to Charge Account.

LD 15 – Add or modify the customer Charge Account.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	CDR	Call Detail Recording Gate Opener.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CHLN	(0)-23	Maximum number of digits that can be entered as a charge account number.

LD 10 – Allow analog (500/2500 type) telephone access to Charge Account.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA	(Deny) allow call transfer.

LD 11 – Allow Meridian 1 proprietary telephone access to Charge Account.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx CPN xx CHG	Add a Calling Party Number key (must be key 24 for the M2317 and key 32 for the M3000). Add a Charge key (must be key 25 for the M2317 and M3000).

LD 12 – Allow attendant console access to Charge Account.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATT 1250 2250	Attendant Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
KEY	0-9 CPN 0-9 CHG	Add a Calling Party Number key. Add a Charge key.

Feature operation

This section explains Charge Account feature and Calling Party Number feature operation for Meridian 1 proprietary telephones, analog (500/2500 type) telephones, and Attendant Consoles.

Meridian 1 proprietary telephones

To charge a call to an account before dialing, follow these steps:

1 Select a free extension.

2 Press **Charge** or dial SPRE + 5.

3 Dial the Charge Account number.

4 When you have a dial tone, dial the telephone number.

To charge during a call in progress, follow these steps:

1 Press **Charge**.

2 Dial the Charge Account number.

3 Press the extension key to return to your call.

To use a SPRE code to charge a call in progress, follow these steps:

1 Press **Transfer** or **Conference**.

2 Dial SPRE + 5.

3 Dial the Charge Account number.

4 Press the extension key to return to your call.

To charge a call to an account when you transfer a call, follow these steps:

1 Press **Transfer**.
The call is on hold.

2 Press **Charge** or dial SPRE + 5.

3 Dial the Charge Account number.

4 Dial the number where the call is to be transferred.

5 Press **Transfer**.

To charge a call to an account when adding a party to a conference call, follow these steps:

1. Press **Conference**.
   The call is on hold.
2. Press **Charge** or dial SPRE + 5.
3. Dial the Charge Account number.
4. Call the party that you want to add to the conference.
5. Press **Conference**.

To record a caller's number for accounting purposes, follow these steps:

1. Press **Calling No**.
   The other party is on hold.
2. Dial a Charge Account number or the caller's number.
3. Press **Calling No.** again to return to the call.

Analog (500/2500 type) telephones

To charge a call to an account before dialing, follow these steps:

1. Select a free extension.
2. Dial SPRE + 5.
3. Dial the charge account number.
4. When you have a dial tone, dial the telephone number.

To charge during a call in progress, follow these steps:

1. Flash the switchhook or link.
2. Dial SPRE + 5.
3. Dial the Charge Account number.
4. Flash the switchhook or link to return to the call in progress.

To charge a call to an account when adding a party to a conference call, follow these steps:

1 Flash the switchhook or link.

2 Dial SPRE + 5.

3 Dial the Charge Account number.

4 Call the party that you want to add to the conference.

5 Flash the switchhook or link.

Attendant Consoles

To charge a call to an account before dialing, follow these steps:

1 Press the **loop** key.

2 Press **Charge**.

3 Dial the Charge Account number.

4 When you have a dial tone, dial the telephone number.

To charge during a call in progress, follow these steps:

1 While the source call is active on a loop key, press **Charge**.

2 Dial the Charge Account number.
The voice connection remains active.

3 Flash the switchhook or link to return to the call in progress.

To record a caller's number for accounting purposes, follow these steps:

1 While the source call is active on a loop key, press **Calling No**.
The other party is on hold.

2 Dial a Charge Account number or the caller's number.

3 Press **Calling No.** again to return to the call.

Charge Account, Forced

Contents

This section contains information on the following topics:

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Feature interactions 967
Feature packaging 969
Feature implementation 969
Feature operation 971

Feature description

Forced Charge Account (FCA) temporarily overrides Class of Service restrictions for toll-denied users. Use Forced Charge Account long distance calls to an account number when calling from a telephone that is restricted from making long distance calls. The unrestricted Class of Service provided by FCA applies for the duration of the call.

When the account number is entered, a charge record is produced on a Call Detail Recording (CDR) device.

FCA supports variable-length numbers of 1 to 23 digits. The minimum value for the account number is specified at the customer level.

A valid account number equals or exceeds the minimum value defined, and is validated by the system for length only. Verification of the actual digits entered is part of Call Detail Recording (CDR) downstream processing.

FCA can be allowed or denied at both customer and user levels. Users include any station or TIE and Common Control Switching Arrangement (CCSA) type trunks assigned a Toll-Denied (TLD), Conditionally Toll-Denied (CTD), or Conditionally Unrestricted (CUN) Class of Service.

Meridian 1 proprietary telephones activate this feature by using a separate Charge key/lamp pair. Any user can access this feature by dialing SPRE + 5.

A distinction is made between normal CDR Charge Account processing and FCA. If all the following conditions are met, the account number is treated as an FCA code:

- The telephone from which the number is entered has a TLD, CTD, or CUN Class of Service.
- The station or trunk from which the number is entered is in a state to originate a call (press a Directory Number [DN] key or flash the switchhook).
- FCA is enabled at the customer level.
- FCA is allowed for the station or trunk from which the number is entered.
- A valid account number is entered at the beginning of the call.

The unrestricted Class of Service provided by FCA, as described above, applies for the duration of the call only. The account number must be reentered for each successive toll call placed by the station or trunk.

Operating parameters

An octothorpe (#) dialed after the account number indicates that the subsequent digits are part of the dialed number.

CDR charge account numbers are fixed-length codes for which a maximum value is specified by the customer. This is also the maximum allowed for the FCA account number.

Because 500 telephones cannot dial an octothorpe (#), they are restricted to fixed-length account numbers.

FCA does not apply to attendant calls.

Feature interactions

Autodial
Speed Call

FCA numbers (including the SPRE code and the Charge Account access code) can be entered in Speed Call lists or stored as Autodial numbers. The digits can also be stored, provided that the account number, regardless of its length, is followed directly by an octothorpe (#).

Authorization Code Security Enhancement

If the Authorization Code is used to change the Class of Service of the user, the new Class of Service must be TLD, CTD, or CUN. If an Authorization Code entered after FCA has altered the Class of Service to unrestricted (UNR), the change made by the Authorization Code still comes into effect.

If the originator's Network Class of Service (NCOS) has been changed by an Authorization Code prior to an applicable FCA entry, the new NCOS is replaced by the FCA NCOS, provided the new Facility Restriction Level (FRL) is not lower than the existing FRL. Similarly, if the originator's NCOS has been changed by an FCA entry, the NCOS will be changed again by a valid Authorization Code entry.

Basic Alternate Route Selection (BARS)

Network Alternate Route Selection (NARS)

If BARS or NARS is equipped, a Network Class of Service (NCOS) associated with FCA must be defined in the Customer Data Block.

Call Detail Recording

Normal Call Detail Recording (CDR) charge account numbers can still be entered before or after an FCA code. If the criteria for an FCA call are not met, (CDR) charge account numbers function in the normal manner.

Call Transfer
Conference

If an FCA code is entered at the beginning of a call, the new unrestricted Class of Service remains in effect for any transfer or conference made during the call. If all FCA criteria are met, an account number entered after activating the Conference key, Call Transfer key, or a switchhook flash is interpreted as an FCA code.

Last Number Redial

These codes are not stored in Last Number Redial (LNR). To use these features when calling the number stored in LNR, the code must first be dialed manually. When dial tone is returned, LNR can be used to complete the dialing.

Pretranslation

The first digit dialed after a valid Charge Account Code is sent to the pretranslator.

Scheduled Access Restrictions

FCA can be used to override Scheduled Access Restrictions (SAR) on a per-call basis, provided the current Class of Service (COS) of the telephone or trunk is CUN, TLD, or CTD. The current COS is the COS in force according to the SAR schedule. If an Authorization Code that sets the COS to CUN, TLD, or CTD is dialed before the FCA, the call is allowed. FCA sets the COS to UNR and the Network COS (NCOS) to the NCOS defined in LD 15, provided that FCA is enabled on both a customer and telephone/trunk basis.

Stored Number Redial

The Forced Charge Account code is not stored. To store a code, dial the code prior to using Stored Number Redial to dial the call.

Trunk Group Access Restrictions (TGAR)

Trunk Group Access Restrictions apply to the telephone or trunk entering the account number.

Feature packaging

This feature is included in base System Software. Forced Charge Account (FCA) package 52 requires:

- Charge Account/Authorization Code Base (CAB) package 24
- Charge Account (CHG) package 23

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Forced Charge Account for a customer.

2 LD 10 – Enable Forced Charge Account for analog (500/2500 type) telephones.

3 LD 11 – Enable Forced Charge Account for Meridian 1 proprietary telephones.

4 LD 14 – Enable Forced Charge Account for each incoming TIE or CCSA trunk.

LD 15 – Enable Forced Charge Account for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	CDR	Call Detail Recording.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CHLN	(0)-23	Maximum number of digits that can be in an FCA code (default is 0).
- FCAF	(NO) YES	(Disable) enable FCA for the customer.

- CHMN	xx	Minimum number of digits that can be in an FCA code (must be less than CHLN).
- FCNC	xx	NCOS to be assigned to FCA codes.

LD 10 – Enable Forced Charge Account for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FCAR	(NO) YES	FCA can be used by this telephone. FCA is restricted from use by this telephone.

LD 11 – Enable Forced Charge Account for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FCAR	(NO) YES	FCA can be used by this telephone. FCA is restricted from use by this telephone.

LD 14 – Enable Forced Charge Account for each incoming TIE or CCSA trunk.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	TIE CAA	Trunk type (must be TIE or CCSA).
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FCAR	(NO) YES	FCA can be used by this trunk. FCA is restricted from use by this trunk.

Feature operation

To use FCA, follow these steps:

1. Select a free extension.
2. Press **Charge** or dial SPRE + 5.
3. Dial the Charge Account number.
4. When you have a dial tone, dial the long distance number.

For operating procedures from particular telephones or consoles, see the Charge Account and Calling Party Number module in this document.

Charge Display at End of Call

Contents

This section contains information on the following topics:

Feature description 973
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Feature interactions 975
Feature packaging 975
Feature implementation 975
Feature operation 976

Feature description

This feature allows the set display of a charged party to show the charged amount of a metered call, along with the normal call-display information. To give you time to read and transcribe the charges, the feature maintains this display for ten seconds after call completion, unless you do something with the set such as make another call or use another feature.

The currency (for instance, pound sterling, mark, and dollar) displayed is the currency specified in the feature configuration (LD 15 and LD 16). The charge information is received from the Central Office (generated from a Periodic Pulse Metering trunk or an Integrated Services Digital Network (ISDN) trunk).

This feature is operational in a standalone environment, and is available on modular digital sets, M2317 sets, and M3000 sets. For M2317 sets and M3000 sets, the charge information is appended to the standard call-display information. On modular sets, the charge information is scrolled to the second line (there are two lines of display on these sets).

When a call is transferred, the Advice of Charge display appears on the set to which the call is transferred. It does not appear on the display of the set that transferred the call.

M2317 and M3000 charge display

For M2317 sets and M3000 sets, the charge information is appended to the standard call-display information. In the example below, the dialed DN (90113145078400) is displayed followed by the cost charged to the call ($22.45).

90113145078400 2245

= = = = =>

Modular Set charge display

On modular sets, which have two lines of display, the charge information is scrolled to the second line. In the example below, the name of the caller is displayed on the first line. The dialed DN (90113145078400), followed by the cost charged to the call ($12.75), is displayed on line two.

LOIS LANE

90113145078400 1275

Operating parameters

The charge is displayed only if all of the following conditions are met:

- the customer to which the set belongs has the Charge Display at End of Call (CHDA) option defined

- the set has a display with Message Registration Allowed Class of Service.
- the trunk is configured with buffered or non-buffered Periodic Pulse Metering.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

The following packages are required for Charge Display at End of Call:

- International Supplementary Features (SUPP) package 131
- Periodic Pulse Metering/Message Registration (MR) package 101

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Implement Periodic Pulse Metering (PPM).

2 LD 15 – Allow or deny Charge Display at End of Call for a customer.

LD 17 – Implement Periodic Pulse Metering (PPM).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System Parameters.
...		
MTRO	PPM	Periodic Pulse Metering

LD 15 – Allow or deny Charge Display at End of Call for a customer.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
OPT	CHDA CHDD	Enter CHDA to allow Charge Display at End of Call. CHDD = Default

Feature operation

No specific operating procedures are required to use this feature.

China – Attendant Monitor

Contents

This section contains information on the following topics:

Feature description 977
Operating parameters 978
Feature interactions 978
Feature packaging 982
Feature implementation 982
Feature operation 983

Feature description

Attendant Monitor Function

Attendant Monitor is a customer defined option which allows the attendant to monitor, in listen only mode, any established call involving a set or trunk on the customer's switch with or without the connected parties being aware that monitoring is taking place (depends on the configuration of the customer tone option).

The differences between the existing Busy Verify and Barge-In features and the Attendant Monitor feature are the following:

- Attendant Monitor provides a listen only path for the attendant.
- There is no click sound given to the connected parties upon attendant connection when the no tone option is configured.

- The tone to the connected parties may or may not be given depending on the new customer tone option for Attendant Monitor.
- The display (if there is one) on any parties involved in the call does not indicate that the attendant is monitoring.

Operating parameters

Since Attendant Monitor is based on Busy Verify and Barge-In, it has the same restrictions which apply to Busy Verify and Barge-in unless otherwise stated in this feature description.

This feature has been developed exclusively for use in China.

Attendant Monitor is strictly a standalone, same-customer feature which is not supported over networks. A customer equipped with the Attendant Monitor feature can only monitor a DN or trunk defined for that customer without going over the network.

The toggle function between having a one-way speechpath and a two-way speechpath during monitoring is not supported.

This feature is not supported for attendants monitoring other attendants.

Monitoring an M2216 or M5317 is not supported.

The attendant may be blocked from monitoring a DN or trunk due to a limit on the number of conference timeslots.

Feature interactions

Attendant Barge-In

When China (CHINA) package 285 is equipped, the normal operation of Barge-In changes slightly. The repeatable tone is now configurable with the (TOA)/TOD option in LD 15.

If an attendant is monitoring a trunk, a second attendant defined at the same customer location is blocked from Barging In to any trunk involved in the monitored call.

If an attendant is Barged-In with a trunk, a second attendant defined at the same customer location will be blocked from monitoring any party involved in the monitored call.

Attendant Break-In

If an attendant is monitoring a DN, a second attendant defined at the same customer site will be blocked from Breaking In to any party involved in the monitored call.

If an attendant is in a Break-In situation with a DN, a second attendant defined at the same customer site will be blocked from monitoring any party involved in the monitored call.

Attendant Busy Verify

When China (CHINA) package 285 is equipped, the normal operation of Busy Verify changes. The repeatable tone is now configurable with the (TOA)/TOD option in LD 15.

If an attendant is monitoring a DN, a second attendant defined for the same customer will be blocked from Busy Verifying any party involved in the monitored call.

If an attendant is Busy Verifying a DN, a second attendant defined for the same customer will be blocked from monitoring any party involved in the monitored call.

Automatic Call Distribution

The attendant cannot monitor a call in which an Automatic Call Distribution (ACD) DN is involved.

Call Forward All Calls
Call Forward Busy
Call Forward, Internal Calls
Call Forward No Answer

If an attendant attempts to monitor a DN which is Call Forwarded and is idle, idle DN treatment is given.

Call Hold, Permanent

Monitoring is not affected if anybody involved in the monitor's call activates hold, except for the case of a simple call. For a monitored simple call, activating hold deactivates monitoring. In all cases, activation of music on hold deactivates monitoring.

An attendant monitoring a call cannot put the monitored DN on hold. The attendant pressing the hold key has no effect while monitoring is enabled.

Call Park

If a DN being monitored becomes parked by another party, the Attendant Monitor feature is deactivated.

Call Trace

If a Call Trace is performed on the attendant, the output will consist of the existing Call Trace information elements for an attendant loop. In addition, "MON" will be printed immediately after "ATTN" to indicate that this attendant is monitoring.

If a Call Trace is performed on any other party involved in the monitored call, the output will consist of the existing information elements for a DN or trunk, as well as "MON" being printed immediately before "ACTIVE".

Call Transfer

If any party at the customer location involved in a monitored call attempts to activate call transfer, monitoring is immediately deactivated.

Centrex Switchhook Flash

If any set at the customer location involved in the monitored call switchhook flashes or performs a Centrex switchhook flash, Attendant Monitor is immediately deactivated.

Conference

If any party involved in a monitored call attempts to activate conference, monitoring is immediately deactivated.With Attendant Monitor active, the attendant cannot create a conference without first disabling the Attendant Monitor feature.

Do Not Disturb

If an attendant attempts to monitor a DN which has Do Not Disturb activated and is idle, idle DN treatment is given.

Make Set Busy

If an attendant attempts to monitor a DN which has Make Set Busy activated and is idle, idle DN treatment is given.

Malicious Call Trace

If a party involved in a monitored call activates the TRC key, monitoring is immediately deactivated.

Multiple Appearance DN

If Attendant Monitor is attempted on a Multiple Appearance DN, the Multiple Appearance Redirection Prime (MARP) TN becomes the desired party.

Override

A set may operate override to join into a desired call. If the desired call is being Attendant Monitored at the time, one of the following occurs:

- If the desired call is a conference call, the override attempt is blocked as per existing operation.
- If the call is a simple one with the Attendant Monitoring with no tone, the override attempt is successful and Attendant Monitor is deactivated.
- If the call is a simple one with the Attendant Monitoring with tone, the override attempt is blocked.

Override, Enhanced

A set may operate enhanced override on a desired call. If the desired call is being Attendant Monitored at the time, existing operation occurs for the first time the Enhanced Override key is pressed. The second time the key is pressed, the interaction with Attendant Monitor is the same as with regular override.

Privacy Release

If Privacy Release is activated on a set that is involved in a monitored call, Attendant Monitor is deactivated.

Private Line

Attendant Monitor is blocked from monitoring a Private DN.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Enable the Attendant Monitor option for a customer, and then either allow or deny the tone option.
2. LD 12 – Define the Barge-In/Attendant Monitor key and the Busy Verify/Attendant Monitor key on an Attendant Console.

LD 15 – Enable the Attendant Monitor option for a customer, and then either allow or deny the tone option.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT	Attendant console options
- OPT	(AMD) AMA	(Deny) allow Attendant Monitor.
	(TOA) TOD	(Allow) deny Attendant Monitor Tone.

LD 12 – Define the Barge-In/Attendant Monitor key and the Busy Verify/Attendant Monitor key on an Attendant Console.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	1250 2250	Attendant Console type.
...		
KEY	00 BVR	Allow both Busy Verify and Attendant Monitor on key 00.
	01 BIN	Allow both Barge-In and Attendant Monitor on key 01.

Feature operation

Monitor a DN

1. Press an idle Loop key on the Attendant Console. The Loop indicator is lit.
2. Press the Busy Verify key. The Busy Verify lamp is lit.
3. Press the Busy Verify key again to enable Attendant Monitor. The Busy Verify lamp is in the flashing state indicating that the Attendant Monitor option is enabled.
4. Dial the extension desired to be monitored. One of the following happens:
   - — Attendant Monitor is blocked when the extension is maintenance busy or disabled.
   - — Attendant Monitor is blocked when the extension is vacant.
   - — Attendant Monitor is blocked when the extension is in some transient state (for example, Conference or Transfer)
   - — Attendant Monitor is blocked when the extension is idle, receiving busy tone, or receiving overflow tone.

— Attendant Monitor is blocked when the extension is involved with another attendant.

— Attendant Monitor is blocked when the extension has activated the Hold key.

— Attendant Monitor is blocked when the extension is already involved in a monitored call by another attendant.

— Attendant Monitor is active when the extension is busy. The attendant is able to listen to all connected parties. If the Attendant Monitor Customer tone is denied (TOD), there is no indication given to the connected parties that the attendant is monitoring. If the Attendant Monitor Customer Tone is allowed (TOA), a burst of tone is sent to the connected parties every 16 seconds. When the monitored DN disconnects from the call, Attendant Monitor is deactivated.

In all of the above situations, the display on the parties involved in the call does not indicate that the attendant has attempted to monitor. With TOD configured, upon the attendant successfully joining the connection there is no click sound given to the monitored parties that could indicate that the attendant has joined in.

5 When Attendant Monitor is deactivated, the attendant is released from monitoring and is free to process calls. Deactivation occurs due to the following:

— The RLS key on the Attendant Console is pressed.

— Any DN involved in the monitored call disconnects.

— Any DN involved in the monitored call at the customer location activates some form of call modification.

Monitor a Trunk

1 Press an idle Loop key on the Attendant Console. The Loop indicator is lit.

2 Press the Barge-In key. The Barge-In lamp is lit.

3 Press the Barge-In key again to enable Attendant Monitor. The Barge-In lamp is in the flashing state indicating that the Attendant Monitor option is enabled.

4 Dial the trunk access code and the route member number, then press "#". One of the following happens:

— Attendant Monitor is blocked when the trunk is disabled.

— Attendant Monitor is blocked when the trunk is idle.

— Attendant Monitor is blocked when the trunk is not assigned.

— Attendant Monitor is blocked when the trunk is already being monitored by another attendant.

— Attendant Monitor is active when the trunk is busy. The attendant is able to listen to all parties on the trunk. If the Attendant Monitor Customer tone is denied (TOD), there is no indication given to the connected parties that the attendant is monitoring. If the Attendant Monitor Customer tone is allowed (TOA), a burst of tone is sent every 16 seconds.

5 When Attendant Monitor is deactivated, the attendant is released from monitoring and is free to process calls. Deactivation occurs due to the following:

— The RLS key on the Attendant Console is pressed.

— The trunk disconnects.

— Any party at the customer location performs some form of call modification or activates hold.

— Any party at the customer location disconnects.

— Any trunk involved in the monitored call disconnects.

China – Busy Tone Detection

Contents

This section contains information on the following topics:

Feature description 987
Operating parameters 988
Feature interactions 988
Feature packaging 989
Feature implementation 989
Feature operation 990

Feature description

In many countries, loop start trunks are not supervised. Therefore, many Public Exchanges/Central Offices send a busy tone to the system when the external party has disconnected. A tone detector must be used to detect this tone.

The China – Busy Tone Detection (BTD) feature allows a technician to enter the characteristics of the busy tone to be detected in overlay 97. This information is downloaded to the system trunk for use during call processing. Once the busy tone is detected, the trunk sends a message to the system software, which then disconnects the call, and the trunk is free for other uses.

An option is provided to allow Busy Tone Disconnect to occur only for incoming calls. This option is also programmed in LD 97. Cadence information is downloaded on a card basis.

Operating parameters

The Busy Tone Detection functionality is implemented on the Enhanced Extended Universal Trunk Card for China (NTRA10AA).

The system will disconnect any call once busy tone is detected on the incoming side of the trunk. If the user on the far end causes busy tone to be generated by any means, the call will be disconnected whether or not that was the intention.

The BTD characteristics are downloaded on a card basis only (not on a unit basis) and thus all units on the trunk card must go to Public Exchanges that produce the same busy tone cadence.

Busy Tone Detection may not work with conference bridges in certain situations. This is due to the nature of conference bridges in such that all of the trunks are incoming. In the situation where two or more loop start trunks with BTD disconnect simultaneously, the resulting busy tone from each trunk may be detected by the BTD inhibitors of the other trunks. The result would be a stalemate where all trunks remain connected.

If a trunk card is not designed to support the Busy Tone Detection feature (for example, for future BTD hardware implementation), the functionality can still be configured in software.

To change Busy Tone Detection assigned to a trunk card, all trunks on the card must first be removed using LD 14.

Feature interactions

Loop Start Supervisory Trunks

The interaction with Intelligent Peripheral Equipment (IPE) trunks occurs because Busy Tone Supervision (BTS) can be configured in conjunction with any existing supervision type. For the EXUT, BTS can now be configured with a supervision type of BST (both incoming and outgoing battery reversal) and Polarity Insensitive (PIP). These supervision type's call processing methods are not changed, except that now the first type of supervision received is the one acted upon.

Feature packaging

China - Busy Tone Detection requires Busy Tone Detection (BTD) package 294, which depends upon Intelligent Peripheral Equipment (XPE) package 203.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 97 – Busy Tone table configuration.

2 LD 14 – Busy Tone Detection Table and Busy Tone Supervision assignment.

For China, the BTD table and its assignment to the various routes is automatic and does not require configuration in LD 97 or LD 14, as long as the BTD package is equipped. Only the configuration of Busy Tone Supervision in LD 14 is required.

LD 97 need only be changed if the values other than the default are required. If table 0 is modified, rather than creating a new one, the assignment to the trunk in LD 14 is not required. After the BTD table is created, or changed, the data must be dumped and the system reloaded in order for the information to be downloaded.

LD 97 – Busy Tone table configuration.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	BTD	Busy Tone Detection.
BTDT	(0)-7 X1-X7	Busy Tone Detection Table.

BCAD	Ph1 Ph2 (350 350)	Busy Tone Cadence. The input values are rounded to the nearest multiple of 25 ms.
BTDD	(BOTH) INC	Busy Tone Detection allowed on both incoming and outgoing calls. Busy Tone Detection allowed on incoming calls only.

LD 14 – Busy Tone Detection Table and Busy Tone Supervision assignment.

Prompt	Response	Description
REQ	NEW	Create or change a route.
TYPE	COT	Central Office trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	EXUT	Extended Universal Trunk.
...		
SIGL	LOP	Loop Start signaling.
...		
SUPN	(NO) YES	Supervision.
STYP	 (PIP) BTS BST PIP BTS BST BTS	Supervision type, where: PIP = Polarity Insensitive BTS = Busy Tone Supervision BST = Both way Supervision Trunk (BST) (that is, PIP on both incoming and outgoing calls) PIP BTS = PIP and BTS, and BST BTS = BST and BTS.
BTDT	(0)-7	Busy Tone Detection Table number.

Feature operation

No specific operating procedures are required to use this feature.

China – Flexible Feature Codes

Contents

This section contains information on the following topics:

Feature description 991
Operating parameters 992
Feature interactions 993
Feature packaging 997
Feature implementation 998
Feature operation 1005

Feature description

Three Flexible Feature Code (FFC) features have been developed to meet the requirements of the Chinese Ministry of Posts and Telecommunications for the rural switch market in China, these features can be used in other markets. The three features are Busy Number Redial (BNR), Customer Call Forward (CCFW), and Outgoing Call Barring (OCB). With all three of these features, there is an option to provide a confirmation tone upon feature activation.

Busy Number Redial

Busy Number Redial enables a user of an analog (500/2500 type) telephone encountering a busy condition to automatically redial the busy number by performing a switchhook flash and dialing the Busy Number Redial FFC. When the user goes off-hook next without dialing any digit, the busy number is automatically redialed. This feature remains in effect until the desired Directory Number (DN) is reached, up to a maximum of 20 minutes.

Customer Call Forward

Customer Call Forward allows users of analog (500/2500 type) telephones to forward their telephones to a central answering position by dialing the Customer Call Forward FFC. This feature activates the Call Forward All Calls function without having to specify the forward DN.

Outgoing Call Barring

Outgoing Call Barring allows a set to be blocked from making some or all outgoing calls. Three levels of barring are available which can be selected by dialing the Outgoing Call Barring FFC, the barring level desired, and the Station Control Password (SCPW).

The three levels are each associated with a New Flexible Code Restriction (NFCR) tree in the Customer Data Block. When a DN is dialed, the digits dialed are compared to the associated NFCR tree and busy tone is given if the call is barred. An FFC is also available to verify that the feature is active.

The active level cannot be changed without first deactivating the feature and reactivating it with a new level.

Operating parameters

Although designed for China, the China – Flexible Feature Codes feature can be used in other markets.

Busy Number Redial is only available for internal calls and for trunk calls that provide a busy signal when busy tone is given.

Busy Number Redial and Customer Call Forward are only available for analog (500/2500 type) telephones.

Outgoing Call Barring does not apply to BRI sets.

The Reply DN for CCFW is limited to 16 digits.

OCB will only process "*" and "#" according to the active NFCR tree if the digits are to be outpulsed on a route with OPR active (including all necessary conditions for OPR). If they are dialed as part of an FFC, the call is allowed; otherwise, an octothorpe will cause the call to be blocked. An "*" will be ignored, except during digit counting. Thus, FFCs containing a "*" or an "#" cannot be blocked by this feature.

OCB will not prevent calls from terminating when there are too few dialed digits to traverse the full NCFR tree (for example, if the active tree is set up to bar 2001, but a DN of 200 exists, calls to 200 will be allowed with no error message).

The maximum number of digits that will be processed by OCB is 32. If the call is not allowed or denied by that point, the call is barred.

OCB can bar feature access codes such as Special Prefix (SPRE) codes and numeric FFCs. It will not bar the digits dialed after a feature access code.

Feature interactions

Busy Number Redial

Autodial

Activation of Busy Number Redial changes the activation of Autodial. The DN that is autodialed becomes the DN that was busy. When the BNR activation timer expires or the busy DN is redialed when it is idle, the autodial capability is deactivated, but the number saved is not cleared. If Autodial is then activated without entering a DN, the number used is the formerly busy DN.

Activation of Autodial when BNR is active deactivates BNR.

Automatic Set Relocation

Busy Number Redial is deactivated when a set is relocated.

Deactivate Feature FFC

The Deactivate Feature (DEAF) FFC deactivates Busy Number Redial.

Hot Line

Busy Number Redial cannot be used on Flexible Hot Line or Enhanced Hot Line sets.

Off-Hook Alarm Security

Busy Number Redial cannot be used on a set with Off-Hook Alarm Security Allowed, since autodial cannot be configured on these sets.

16-Button Dual-tone Multifrequency (DTMF) Sets

Busy Number Redial (BNR) activate can be a postdial function, and BNR deactivate can be a predial function. Both FFCs may be dialed normally from a 16-button DTMF telephone.

Customer Call Forward

Customer Call Forward (CCFW) is another way of activating Call Forward All Calls (CFWAC), therefore all interactions with CCFW not specified below are the same as with CFWAC.

Call Forward All Calls

When CCFW is active Call Forward All Calls (CFWAC) cannot be activated by Flexible Feature Code, but can be activated by SPRE. When CFWAC is active, CCFW cannot be activated.

CCFW can be deactivated by deactivating CFWAC. CFWAC can only be deactivated by the CCFD FFC if the current CFW DN is the same as the current CCFW DN.

Deactivate Feature FFC

The Deactivate Feature FFC deactivates Customer Call Forward.

Make Set Busy

Customer Call Forward takes precedence over Make Set Busy if both are active.

16-Button Dual-tone Multifrequency (DTMF) Sets

CCFA and CCFD are allowed as predial ABCD functions. They may also be dialed normally from 16-Button DTMF telephones.

Outgoing Call Barring

Outgoing Call Barring is an additional feature that may block a call. All other call blocking features still apply as usual.

Authorization Code Security Enhancement

Digits dialed after an Authorization Code are checked against the active OCB level.

Call Forward All Calls

When a set with Outgoing Call Barring active activates CFWAC with a new CFW DN, the CFW DN is checked against the current barring level. If the DN is not allowed to be dialed, it can also not be used as a Call Forward DN. This is to prevent a set from forwarding to a barred DN and then dialing its own DN to bypass the restrictions.

Charge Account and Calling Party Number

Digits dialed after a charge account are checked against the active OCB level.

Digit Display

Meridian 1 proprietary telephones with displays do not display the OCB level and the Station Control Password (SCPW) when OCB FFCs are dialed. This protects the security of the SCPW.

Flexible Feature Codes

Flexible Feature Codes containing a “*” or an “#” will always be allowed by OCB. Therefore, FFCs which can be used to make a call should be entirely numeric if barring of them is required.

Some FFCs are equivalent to Special Prefix functions and these will be subject to barring based on the equivalent Special Prefix codes, even if the FFC is entirely numeric.

Last Number Redial

Barred DNs will be stored by Last Number Redial (LNR). DNs redialed using LNR are checked against the active OCB level.

OCB Flexible Feature Codes are not stored as the last number dialed.

Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS)

Calls made through NARS or BARS that result in local termination may not be barred based on the full dialed digits. They will be barred based on the resulting local digits. Calls to a Local Steering Code will use the full dialed digits, as well as the resulting termination. Calls to the Home Location Code (HLOC) will be barred based on the AC1 and HLOC, and then barred separately based on the local DN. Other local terminations include calls to an Numbering Plan Area (NPA) or Home Numbering Plan Area (HNPA) that use Supplemental Digit Restriction and Recognition to recognize a Local Direct Inward Dialing (DID) number, and calls using a Route List Block which terminate using the LTER option.

New Flexible Code Restriction

Outgoing Call Barring uses New Flexible Code Restriction (NFCR) trees to define the digit sequences that are not allowed for each level of barring. However, OCB analyses all dialed digits, whereas NFCR only analyses digits outpulsed on trunks. This means that the same tree will not normally be usable for both features, unless only Coordinated Dialing Plan trunk calls are to be blocked for both features and no digit manipulation is done.

Outpulsing of Asterisks and Octothorpes

The NFCR trees used for Outgoing Call Barring are allowed to contain asterisks and octothorpes when the Outpulsing of Asterisk and Octothorpe (OPAO) package 104 is equipped. These special digits will only be used for processing dialed digits on routes with OPAO allowed.

Remote Call Forward

Activation of CFW to a barred DN by Remote Call Forward will be permitted, since the user has had to dial the Station Control Password, which could also have been used to deactivate OCB.

Ring Again

Ring Again cannot be activated after a call is barred by Outgoing Call Barring. Sets with display will not offer Ring Again.

Saved Number Redial

DNs redialed using Saved Number Redial (SNR) are checked against the active OCB level.

M2317 sets will offer a “Save #” after a call to a barred DN.

Speed Call

Digits dialed using Speed Call are checked against the active OCB level. This includes calls made using the Dial Access to Speed Call feature (that is, using Pilot DNs).

Speed Call, System

Digits dialed using System Speed Call are checked against the active OCB level.

16-Button Dual-tone Multifrequency (DTMF) Sets

The Outgoing Call Barring FFCs are not allowed as ABCD functions. They may be dialed normally from 16-Button DTMF telephones.

Feature packaging

China – Flexible Feature Codes is contained in base system software. The following packages are required for feature activation:

- Flexible Feature Codes (FFC) package 139
- Busy Number Redial also requires the Autodial functionality that is contained in Optional Features (OPT) package 1
- Customer Call Forward also requires the Call Forward All Calls functionality contained in Optional Features (OPT) package 1 and Special Service for 2500 Sets (SS25) package 18
- Outgoing Call Barring also requires the New Flexible Feature Code Restriction (NFCR) package 49

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Define the basic setup for all three of the FFC-based features.
2. LD 56 – Define Confirmation Tone.

LD 15 – Define the basic setup for all three of the FFC-based features.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FFC	Flexible Feature Codes
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- FFCS	(NO) YES	(Do not) change Flexible Feature Code end-of-dialing indicator.
- STRL	1-3	String Length of end-of-dial indicator.
- STRG	xxx	String to indicate end-of-dialing. Up to three digits as defined by STRL. Digits 0-9 and “#” are valid entries.
REQ	CHG	Change.
TYPE	FTR	Features and options.
- SPRE	nnnn	Special Function Prefix (required for FFCs to operate).

LD 56 – Define Confirmation Tone.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTC	Flexible Tones and Cadences.

TABL	0-31	FTC Table Number.
HCCT	YES	Hardware controlled cadences and tones.
...		
- FFCT		Configure Confirmation Tone.
- - XTON	0-(4)-255	XCT (NT8D17 Conference/TDS) Tone code.
- - XCAD	(0)-255	XCT (NT8D17 Conference/TDS) Cadence number (FCAD cadence number).

Busy Number Redial

Task Summary List

Complete the following steps to configure the Busy Number Redial feature:

1 LD 57 – Configure Flexible Feature Codes for Busy Number Redial.

2 LD 15 – Set the Autodial Delay time used by Busy Number Redial.

3 LD 10 – Allow Busy Number Redial for analog (500/2500 type) telephones.

LD 57 – Configure Flexible Feature Codes for Busy Number Redial.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
CODE		Change code.
- DEAF	xxxx	Deactivate Feature.

- BNRA	xxxx	Busy Number Redial activation code.
- BNRD	xxxx	Busy Number Redial deactivation code.

LD 15 – Set the Autodial Delay time used by Busy Number Redial.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FFC	Flexible Feature Codes
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- ADLD	(0)-20	Number of seconds to delay before autodialing the saved busy number (0 will make the feature unavailable). Odd entries are rounded up to the next even number.

LD 10 – Allow Busy Number Redial for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(BNRD) BNRA	(Deny) allow Busy Number Redial for this telephone.
FTR	ADL xx	Autodial, where: xx = the maximum number of digits that can be stored.

Customer Call Forward

Task summary list

The following is a summary of the tasks in this section:

1 LD 57 – Configure Flexible Feature Codes for Customer Call Forward.

2 LD 15 – Add or change the Reply DN for Customer Call Forward.

3 LD 10 – Configure the Call Forward feature to allow Customer Call Forward for analog (500/2500 type) telephones.

LD 57 – Configure Flexible Feature Codes for Customer Call Forward.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
CODE		Change code.
- DEAF	xxxx	Deactivate Feature.
- CCFA	xxxx	Customer Call Forward activation code.
- CCFD	xxxx	Customer Call Forward deactivation code.

LD 15 – Add or change the Reply DN for Customer Call Forward.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDR	Call Redirection

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CCFWDN	dd..dd X	Customer Call Forward DN (maximum of 16 digits). X to delete.

LD 10 – Configure the Call Forward feature to allow Customer Call Forward for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FTR	CFW nn	Call Forward (nn must be the same number of digits as the CCFW DN).

Outgoing Call Barring

Task summary list

The following is a summary of the tasks in this section:

1. LD 57 – Configure the Flexible Feature Codes for Outgoing Call Barring.
2. LD 15 – Specify NFCR trees and Station Control Password length for Outgoing Call Barring.
3. LD 49 – Create NFCR trees for Outgoing Call Barring.
4. LD 10 – Allow Outgoing Call Barring for analog (500/2500 type) telephones.
5. LD 11 – Allow Outgoing Call Barring for Meridian 1 proprietary telephones.

LD 57 – Configure the Flexible Feature Codes for Outgoing Call Barring.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
CODE		Change code
- OCBA	xxxx	Outgoing Call Barring activation code.
- OCBD	xxxx	Outgoing Call Barring deactivation code.
- OCBV	xxxx	Outgoing Call Barring verification code.

LD 15 – Specify NFCR trees and Station Control Password length for Outgoing Call Barring.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FCR	New Flexible Code Restriction options
NFCR	(NO) YES	(Disable) enable New Flexible Code Restriction.
- MAXT	1-255	Maximum number of NFCR trees.
- OCB1	0-MAXT	NFCR tree for Outgoing Call Barring level 1. Enter 255 to deactivate.
- OCB2	0-MAXT	NFCR tree for Outgoing Call Barring level 2. Enter 255 to deactivate.
- OCB3	0-MAXT	NFCR tree for Outgoing Call Barring level 3. Enter 255 to deactivate.
TYPE	FFC	Flexible Feature Codes

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- SCPL	0-8	Station Control Password Length.

LD 49 – Create NFCR trees for Outgoing Call Barring.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	FCR	Flexible Code Restriction.
CUST	xx	Customer number, as defined in LD 15
CRNO	(0)-254	New Flexible Code Restriction tree number.
INIT	ALOW DENY	Allow or deny digit strings not in tree.
ALOW	xxxx	Digit sequence to be allowed unconditionally.
	xxxx y...y	Digit sequence to be conditionally allowed and maximum number of digits that can follow.
UPDT	(YES), NO	Data is correct and NFCR tree can be updated.
DENY	xxxx	Digit sequence to be denied unconditionally.
	xxxx y...y	Digit sequence to be conditionally denied and maximum number of digits that can follow.
UPDT	(YES) NO	Data is correct and NFCR tree can be updated.

LD 10 – Allow Outgoing Call Barring for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCPW	xxxx	Station Control Password. Length must match SCPL in LD 15.
CLS	(OCBD) OCBA	(Deny) allow Outgoing Call Barring.

LD 11 – Allow Outgoing Call Barring for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCPW	xxxx	Station Control Password. Length must match SCPL in LD 15.
CLS	(OCBD) OCBA	(Deny) allow Outgoing Call Barring.

Feature operation

Busy Number Redial

To activate Busy Number Redial, a switchhook flash is done, and the BNRA FFC is dialed. The number to be redialed is stored in the set's storage area for Autodial. Confirmation tone is given if configured.

When the user goes off-hook and does not dial within the number of seconds configured for the Autodial Delay timer (ADLD) the busy number is automatically redialed. If the DN is idle when redialed, the feature is deactivated, otherwise it remains available for 20 minutes. The feature can be deactivated manually by dialing the BNRD FFC, or the Deactivate Feature FFC. The user again hears confirmation tone. Deactivating the feature when it is not active will still cause confirmation tone to be heard, since the feature may have just timed out when going off-hook to dial the deactivation code.

Customer Call Forward

Customer Call Forward is activated by dialing the CCFA FFC. Confirmation tone is given if activation is successful. When Customer Call Forward is activated, the CFWAC feature is activated with the CFW DN set to the Reply DN.

CCFW is deactivated by dialing the Customer Call Forward Deactivate FFC, or the Deactivate Feature FFC (it will also be deactivated by the activation of certain other features, see the Feature interactions section of this description).

Outgoing Call Barring

Outgoing Call Barring can be activated from a set with OCBA Class of Service. To activate OCB, the subscriber dials the OCBA FFC, the barring level desired, and the Station Control Password. An octothorpe must also be dialed from Meridian 1 proprietary telephones and analog (500/2500 type) telephones. If activation is successful confirmation tone is given. If the feature is not activated overflow tone is given.

The feature is deactivated by dialing the OCBD FFC, then the active level, and the Station Control Password. An octothorpe must also be dialed from Meridian 1 proprietary telephones and analog (500/2500 type) telephones. The subscriber may dial the OCBV FFC to verify that OCB is active. Confirmation tone is heard if OCB is active, otherwise overflow tone is heard.

China – Supervised Analog Lines

Contents

This section contains information on the following topics:

Feature description 1007
Operating parameters 1008
Feature interactions 1010
Feature packaging 1011
Feature implementation 1011
Feature operation 1012

Feature description

The China – Supervised Analog Lines feature provides two types of call supervision signaling capabilities: battery reversal answer/disconnect supervision and hook flash disconnect supervision. These forms of supervision are provided to terminal devices connected to analog ports in the system.

Battery Reversal Supervision

Battery reversal answer and disconnect supervision signaling is used for calls originating from the terminal device. It provides both far-end (the called party) answer supervision and far-end disconnect supervision signals to the terminal device. It does not apply to incoming calls terminating at the terminal device.

In the idle state, the analog port in the system provides ground signal on the tip lead and battery on the ring lead. This polarity is maintained during dialing and ringing at the far end. When the far end answers, the battery and ground connections are reversed. The reverse battery is maintained while the call is established. When the far end disconnects, the battery and ground connections are reverted to the idle state to signal that the far end has disconnected. If the terminal device disconnects first, the system sends the Deactivate Battery Reversal Scan Signal Distribution (SSD) message to the firmware after receiving the on-hook status to revert the polarity to its idle state.

Two types of battery reversal are supported. Battery Reversal for Absolute Answer Only provides an answer supervision signal to the terminal device only when the system detects an absolute answer. Battery Reversal for Absolute and Assumed Answer provides an answer supervision signal to the terminal device even when an assumed answer is detected and the far end is not capable of indicating definite answer (for example, an outgoing call on an unsupervised loop start trunk).

Hook Flash Disconnect Supervision

Hook flash disconnect supervision is used for incoming calls terminating at the terminal device. The disconnect signal is indicated by the removal of the ground connection to the tip lead for a specific period of time, which is provided by firmware ranging from a minimum of 10 milliseconds to a maximum of 2.55 seconds. The analog port is held busy for incoming calls while hook flash is in progress.

Operating parameters

This feature applies to Intelligent Peripheral Equipment that support the Supervised Analog Line feature only.

Disconnect supervision is not provided to the terminal device if the system does not receive any indication of the far end releasing.

If the system does not receive any answer indication, and answer supervision is not extended to the terminal device following an assumed answer condition, disconnect supervision cannot be extended when the far end disconnects.

If the Battery Reversal Supervision feature is configured for an analog line on an analog card that does not support battery reversal, the battery reversal SSD messages from the system software are ignored by the analog card firmware. In this case, no battery reversal signal is extended to the terminal device.

If the Hook Flash Disconnect Supervision feature is configured for an analog line on an analog card that does not support hook flash, the hook flash SSD messages from the system software are ignored by the analog line card firmware. In this case, no hook flash signal is extended to the terminal device.

If the system initializes while an outgoing call originating from an analog line is established and battery reversal is activated, unprotected data for the call is lost. In this case, battery reversal remains activated when the call is cleared down by either party. However, the line status is reverted to normal when the next outgoing call is answered and then cleared down.

If the hook flash timer is set equal to or greater than the on-hook timer, activation of the hook flash disconnect signal also causes the card to send an on-hook message and then an off-hook message to the system. In this case, if the user remains off-hook after the far end disconnects, dial tone is received and an outgoing call can be initiated.

Battery reversal supervision is supported on the following cards:

- The Enhanced Extended Analog Line Card for China NTRA08AA or later vintage.
- The Off-Premise Station Line Card NT1R20AB or later vintage, and
- The Off-Premise Station Line Card for China NTRA06AB or later vintage.

Hook flash disconnect supervision is supported on the following cards:

- The Off-Premise Station Line Card NT1R20AB or later vintage, and
- The Off-Premise Station Line Card for China NTRA06AB or later vintage.

Feature interactions

Call Transfer

If more than one active call is extended to an analog line, the call type associated with an analog line is determined by the first active call. The call type is assumed to be incoming and hook flash supervision applies if a terminal device answers an incoming call from an idle state. If the terminal device performs a switch hook flash to put the first party on hold and initiates a consultation call, the Battery Reversal feature is not supported; no battery reversal answer signal is extended to the terminal device when the second party answers.

If the first party disconnects while the terminal device is connected to the second party, no disconnect supervision is extended to the terminal device. However, hook flash disconnect supervision is extended to the terminal device when the second party disconnects (that is, a disconnect supervision signal is sent only when the last party connected to the terminal disconnects).

If a terminal device originates an outgoing call, battery reversal answer supervision is extended when the called party answers. The polarity of the line remains reversed when the terminal device performs a switch hook flash and then initiates a consultation call to a second party. The analog line is reverted to normal polarity when the terminal device completes the transfer and drops out or when the last of either the held party or the consultation party disconnects.

Conference

If a terminal device answers an incoming call and then initiates a conference, no battery reversal answer supervision signal is extended to the terminal device when new parties of the conference answer. However, a hook flash disconnect supervision signal is extended to the terminal device when the last party in the conference disconnects.

If a terminal device initiates a conference, battery reversal answer supervision is extended to the terminal device when the first party answers. No polarity change is made when additional parties are added to the conference. The polarity is reverted to normal when the terminal device disconnects or when the last party in the conference disconnects.

Multi-Party Operations

As in the cases with Call Transfer and Conference, the call type of the first active call determines whether battery reversal or hook flash supervision applies. Also, supervision signaling is not supported for the second call. A disconnect supervision signal is extended only when the last party disconnects.

Feature packaging

This feature is included in base System Software

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable battery reversal supervision.

2 LD 10 – Enable hook flash disconnect supervision..

LD 10 – Enable battery reversal supervision.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

...		
FTR	OSP (1)	Outgoing call supervision. Answer and disconnect supervision for outgoing calls with absolute and assumed answer indication. If the numeric parameter is not entered and the saved value is null, it is defaulted to 1. Otherwise it remains unchanged.
	OSP 2	Answer and disconnect supervision for outgoing calls with absolute answer supervision only.
	XOSP	Enter XOSP to disable battery reversal answer and disconnect supervision.

LD 10 – Enable hook flash disconnect supervision.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
FTR	ISP 1...(75)...255	Enable hook flash disconnect supervision with flash timer in 10 millisecond units. If the numeric parameter is not entered and the saved value is null, it is defaulted to 75. Otherwise it remains unchanged.
	XISP	Enter XISP to disable hook flash disconnect supervision.

Note: Respond to the FTR prompt in LD 10 with OSP 1, and then with ISP 1...(75)...255 to enable both battery reversal supervision and hook flash disconnect supervision.

Feature operation

No specific operating procedures are required to use this feature.

China – Toll Call Loss Plan

Contents

This section contains information on the following topics:

Feature description 1013
Operating parameters 1014
Feature interactions 1014
Feature packaging 1015
Feature implementation 1016
Feature operation 1018

Feature description

The China – Toll Call Loss Plan feature provides the requirement of 7 dB loss for 2.0 Mbps Digital Trunk Interface (DTI2) toll calls from a system, acting as a Class 5 (C5) office. It is only applicable for a DTI2 trunk connection using MFC signaling to an analog (500/2500 type) telephone.

Feature specific losses on the DTI2 card and 500/2500 line card are provided if the call is recognized as a toll call and the local party is using an analog (500/2500 type) telephone. The loss levels for toll calls are configured in LD 73 by using the TOLT and TOLL prompts.

With this feature, when a toll condition is detected, loss levels are sent to the ONS line card. When the call is terminated, the original loss levels are sent to the ONS line card.

The outgoing toll call is recognized by defining the toll digits as a Special Service List number in LD 18 and specifying it in the DTI2 Route Data Block. For incoming calls, the toll status is provided by the Multifrequency Compelled (MFC) signaling. When the toll status is determined, the appropriate pad values are used on the DTI2 card and 500/2500 line card.

Operating parameters

The Toll Loss Plan is only supported when a pure Intelligent Peripheral Equipment (IPE) Loss Plan for China is used. Existing pad levels for DTI2 toll calls with Hybrid Loss Plan for China are unchanged.

This feature is only supported for a Class 5 system.

The Toll Loss Plan is only supported on DTI2 MFC trunks.

If an initialization occurs, the toll call stays connected, but the toll status is lost. The pad levels are reinserted as if it is for a non-toll call.

The Toll Loss Plan is not supported when a conference call is in progress.

China specific IPE 500/2500, DTI2, and MFC line cards are required.

Feature interactions

Call Diversion (Call Transfer, Call Forward All Calls, Call Forward Busy, Call Forward, Internal Calls, Call Forward No Answer, Hunting)

Toll pad switching is also provided after call diversion has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Conference

Toll Loss Plan is not supported when a conference is in progress. When a local party connecting to a toll call makes a conference call, the pad levels on the ONS line card are switched back to their original (non-toll call) values. Then, the existing Conference algorithm takes care of the necessary pad switching. This would not alter the existing conference call in terms of loss levels.

When a conference call joins in a toll call, the Toll Loss Plan is not effective.

When a conference call involving a toll call becomes a two-party call, the Toll Loss Plan is applied on the set and DTI2 trunk.

The conference pad switching algorithm is not changed for the Toll Loss Plan, since the 7 db requirement does not apply to a Conference call.

Digitone Receiver Pads

When a Digitone Receiver (DTR) is connected to the DTI2 call (receive only), the pad value to be used on the DTI2 pack is defined in LD 73. There is no interaction with this since the Toll Loss Plan is only supported on a DTI2 trunk with MFC signaling.

Multifrequency Compelled Signaling Pads

There is no interaction with this operation, since the Toll pad switching is only performed when the call is established and the Multifrequency Compelled signaling has terminated.

Multi-Party Operations

When a user toggles between one party and another, the Toll Loss Plan is inserted on the active call if it is a toll call. If the user toggles to a non-toll call, the Toll Loss Plan is removed.

Feature packaging

The China Toll (CHTL) package 292 must be provisioned to activate this feature.

In addition the China – Toll Call Loss Plan feature requires:

- Multifrequency Compelled Signaling (MFC) package 128
- 2 Mbps Digital Trunk Interface (DTI2) package 129
- International Supplementary Features (SUPP) package 131 for the Special Service List functionality.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 18 – Define toll digits for China.

2 LD 16 – Enter the SSL number (defined in LD 18) in the Route Data Block of the DTI2 trunk.

3 LD 73 – Enter the pad codes for a toll call on DTI2.

LD 18 – Define toll digits for China.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	SSL	Special Service List.
SSL	1-15	Enter Special Service List number.
SSDG	0	China national toll call.
- TOLL	YES	The SSDG entry is a toll number.
...		
SSDG	00	China international toll call.
- TOLL N	YES	The SSDG entry is a toll number.

...		
SSDG	<CR>	End entry.

LD 16 – Enter the SSL number (defined in LD 18) in the Route Data Block of the DTI2 trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block
...		
SSL	1-15	Enter the SSL number defined in LD 18.

LD 73 – Enter the pad codes for a toll call on DTI2.

Prompt	Response	Description
REQ	NEW CHG PRT	Add, change, or print.
TYPE	DTI2	2.0 Mbps DTI data block.
FEAT	PAD	Pad category.
PDCA	1-16	Pad category table.
TNLS	(NO) YES	Terminal Number list.
DFLT	(1)-16	Default table.
...		

TOLT	Rx Tx (0) (0)	Toll call pad data on DTI2 card, where: Rx = Receive, and Tx = Transmit. The default values are 0 dB receive, and 0 dB transmit.
TOLL	Rx Tx (16) (30)	Toll call pad data on line card, where: Rx = Receive, and Tx = Transmit. The default values are 0 dB receive, and 7 dB transmit.

Feature operation

No specific operating procedures are required to use this feature.

CIS ANI Digits Manipulation and Gateways Enhancements

Contents

This section contains information on the following topics:

Feature description 1019
Operating parameters 1032
Feature interactions 1033
Feature packaging 1035
Feature implementation 1036
Feature operation 1052

Feature description

This chapter describes the Commonwealth of Independent States (CIS) Automatic Number Identification (ANI) Digits Manipulation and Gateways Enhancements features. The features are presented together because they share many common characteristics.

CIS ANI Digits Manipulation and Gateways Enhancements features allow the ANI to be built in a more flexible way when the call is originated from a set and from a route.

ANI Definition

The Automatic Number Identification (ANI) information is a string of digits sent to the Central Office (CO), which it uses to identify the calling subscriber for billing purposes, Malicious Call Trace (MCT) purposes, and for immediate information about the subscriber when reaching some vital service such as fire brigade, emergency medical care, or law enforcement officials. The ANI information is sent over the speech path whenever the CO requests it.

ANI is sent on the following CIS trunks on the system:

- CIS three-wire analog trunk
- CIS digital trunk interface Dial Pulse (DP) and Multi-Frequency Shuttle (MFS)

Shared functionality

When configuring outgoing CIS routes, the feature is activated using the prompt ANIC to enable composition of a new ANI.

Two similar ANI tables are configured in the customer data block. One is used when call is originated from a set. The second table is used when call is originated from a route.

Up to 2000 entries can be configured in the ANI table for sets. Up to 512 entries can be configured in the ANI table for routes.

An ANI entry contains the following data:

- DNLG (DN Length): it gives the number of digits of the calling number that will be used for building the ANI DN (for example calling party DN = 342390 and DNLG = 4 will give an ANI DN = 2390).
- LEC: Local Exchange Code.
- ANDN: used as ANI DN if DNLG=0.
- ADDG: additional digits.

CIS ANI Digits Manipulation introduction

The CIS ANI Digits Manipulation feature enhances the system CIS ANI digits manipulation capabilities by increasing flexibility and ensuring greater accuracy. The feature modifies the length of outgoing ANI and it allows the building of outgoing ANI using the table configured in customer data block.

CIS Gateway Enhancements introduction

The purpose of the gateways enhancement feature is to enhance several gateways with Commonwealth of Independent States (CIS) trunks.

CIS gateways considered are with all route types with an emphasis on connectivities that can receive a Calling Number: R2MFC, ISDN interfaces and incoming CIS DTI2.

ANI Digits Manipulation feature functionality

The ANI may be built in two ways with the enhancements introduced by the CIS ANI Digits Manipulation feature as described in the list below (see Figure 14):

1 The ANI may be built with the following modifications:

— The length of the ANI information built by the software is configurable on a per route basis and may reach 15 digits (for LEC+ANI DN).

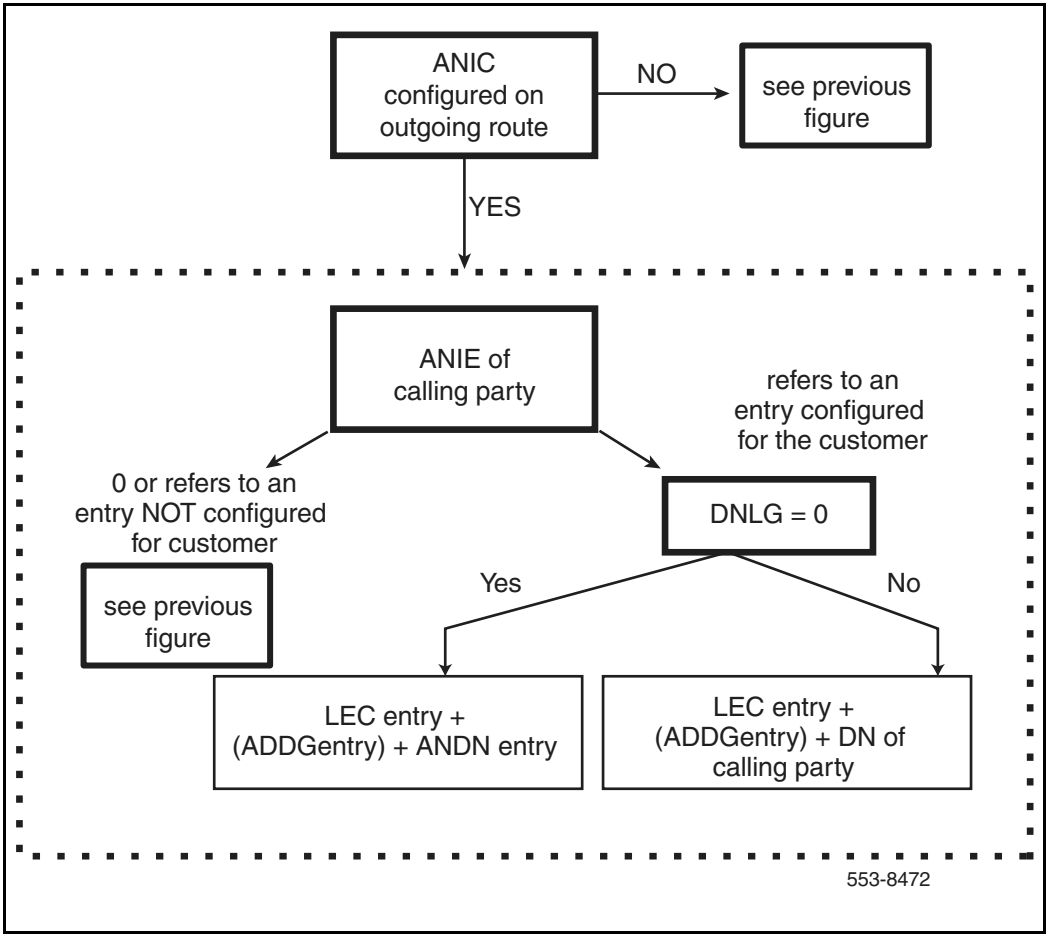
— The part of the ANI DN to be truncated (in case the truncation can't be avoided) is the beginning (that is the most significant digits).

— The system has the option to work without LEC, that is the response to the prompt LEC in LD 16 is not mandatory anymore.

— In the case of LEC+ANI DN smaller than the programmed length, additional digit(s) (ADDG) is (are) added at the beginning of the ANI DN (between the ANI DN and the LEC), in compliance with the CIS standards. ADDG can be more than one digit long.

2 Optionally, the ANI data may be retrieved from entries configured in LD 15. It provides much more flexibility in building the ANI. An ANI entry number can be assigned to each circuit switched network set, BRI set and BCS DN key.

Figure 14
Example of how ANI is built in call originating from a set (using enhanced functionality)



CIS ANI Digits Manipulation examples

This section provides diagrams and tables that show how CIS ANI Digits Manipulation feature operates. Example 1 is presented in Table 34. Example 2 is shown in Table 35.

Example 1 - DN on key 1 is to be used in ANI

Table 34
DN on key 1 is to be used in ANI

ANI entry	Entry X	Entry Y
DNLG, length of DN to use in ANI	4	0
LEC, Local Exchange Code	940	940
ADDG, additional digits	89	8
ANDN, used as ANI DN if DNLG is 0		7676

Example 2 - DN on key 1 is not to be used in ANI

Table 35
DN on key 1 is not to be used in ANI

ANI entry	Entry X	Entry Y
DNLG, length of DN to use in ANI	4	0
LEC, Local Exchange Code	940	940
ADDG, additional digits	89	8
ANDN, used as ANI DN if DNLG is 0		7676

Gateways Enhancement feature functionality

The gateway enhancements are composed of the following new functionalities, which are described in this section:

- Mapping of the received calling number to ANI:
  - — Mapping CNI to ANI, ANI to CNI
  - — Mapping CLID to ANI and OLI to ANI
  - — Mapping ANI to ANI in gateway CIS to CIS
- Building ANI for interfaces without calling number

Mapping of the received calling number to ANI

To map any Calling number to ANI the idea is to be able to manipulate the received Calling Number when tandeming it into ANI. In a gateway situation the incoming trunk is the originator of the outgoing call. So the Calling Number is manipulated by assigning an ANI table entry to the incoming route. An ANI table entry is also assigned to route types whose connectivity does not give the possibility to receive a Calling Number. For these routes, the ANI table entry allows to build the outgoing ANI using only information pertaining to the incoming route.

CIS Gateway Enhancement introduces a list of ANI entries applicable to incoming route considered for the gateway with CIS, or incoming and outgoing route. When package CIST is equipped, an ANI entry is attached to each applicable route. An entry for routes is configured with prompt: R_ENTRY.

Some of the fields defined in ANI table entry were already existing in route configuration. ANI table entry fields that were already existing on route (RDNL, ANDN) are suppressed from the route. It must be noticed that prompt RDNL of Route Data Block is replaced by prompt DNLG in ANI table entry. A new prompt is added on Route data configuration to defined ANI table entry: ANIE. During upgrades, former Route Data Block prompt values are moved into an ANI table entry and this table entry is assigned to the route.

Several routes can have the same tandeming information configured. The data has been moved from Route Data Block to Customer Data Block. In case no ANI table entry is configured, the default table entry (0) is used.

ANI is built as follows:

If outgoing route does not allow use of ANI table entry assigned on incoming route (ANIC prompt of outgoing CIS route is set to NO), Calling number is built as before. Notice that part of the received Calling Number (DNLG) and ANDN INC route are configured in ANI table entry of the incoming route.

1 If outgoing route allows to use ANI table entry assigned on incoming route (ANIC is set to yes on CIS outgoing route), ANI is built as LEC inc route + ADDG inc route + (part of) received Calling Number or, when calling Number not available.

Note: With new implementation "part of received calling Number" indicates N least significant digits of received Calling Number, where N is the value configured against prompt DNLG in the ANI table entry associated to the incoming Route Data Block.

2 LEC inc route + ADDG inc route +ANDN inc. route

ANI has a predetermined size, configured on outgoing CIS route. It can happen that ANI built is smaller or longer than the ANI configured size. This is handled the same way as for ANI built when call originator is a set.

When the ANIC of the outgoing route is set to yes, data from the ANI table entry configured for the incoming route is used, even if empty. Thus it is up to the craftsman to ensure that the ANI built using the ANI table entry is correct.

Mapping CNI to ANI, ANI to CNI

On MFC routes, prompt ANDN is currently defined. This prompt is removed and replaced by prompt ANIE which allows configuring the ANI table entry number.

Both CNI and ANI contain a CAC field. The value and meaning of the field in both cases are not the same.

On the gateway side of the R2MFC to CIS, the purpose of this mapping is to convert the 10 MFC CAC meanings into CIS CAC meanings, or to define a default CIS CAC value. For this, a conversion table is defined. Conversion tables are defined in LD 15. Up to 32 tables can be configured. A conversion table can be assigned on each MFC route. For this, the prompt CAC_CONV is added to the MFC incoming route configuration. If no CAC conversion table is configured on the MFC route, default table (number 0) will apply to the MFC incoming route.

A CAC conversion entry contains the following data:

- CAC0 to CAC9: against CAC0 prompt is configured the CIS CAC value to be sent if an R2MFC CAC of value 0 has been received.
- DFLT: against this prompt is configured the CIS CAC value to be sent if the R2MFC CAC has not been received or is not in the correct range.

When CNI has not been received at the time ANI is built, a default value must also be used. This default value is defined in the CAC Conversion table. On CIS trunks, 10 CAC values can be sent, from 0 to 9. Each one has a particular meaning (see Table 36).

Default table is number zero and is configurable. After upgrade this table is defined in memory as shown in table below. The craftsman must be aware that modifying this default table after conversion will modify the CAC generated for all gateway cases where the incoming route had a CAC set to three (before upgrade).

In this default table all fields are set to three, as it is the current CIS CAC default value. A value of three for CIS Category Code means the caller is residential, business or hotel subset with the access to local network and without access to automatic regional, toll, international network and to chargeable service numbers.

Table 36
CAC conversion table entry 0 for R2MFC route

MFC CAC	CIS CAC	CIS CAC description
CAC0	3	CIS value corresponding to MFC CAC DGT0
CAC1	3	CIS value corresponding to MFC CAC DGT1
....	3	
CAC8	3	CIS value corresponding to MFC CAC DGT8
CAC9	3	CIS value corresponding to MFC CAC DGT9
Default	3	CIS value used when MFC CAC has not been received, or MFC CAC received is not in the MFC CAC list of this table

MFC routes were given a CAC prompt on which was configured the CAC value of incoming route when tandeming to CIS. This data has no more utility on incoming MFC route as a CAC conversion table is configured against prompt CAC_CONV. Thus, CAC prompt is removed from MFC incoming routes. On upgrade, former CAC prompt value is moved into a CAC conversion table, and this table is assigned to the route.

In the gateway side CIS to R2MFC, the purpose of this mapping is to convert the 10 CIS CAC meanings into R2MFC CAC value. For this, a conversion table is defined.

Prior to upgrading, the CAC sent was defined in the R2MFC table (LD 94). In this R2MFC table, it is the same value for all non-tie incoming trunks. In LD 15, for CAC conversion table, the same range (1-10) and default value (6) than in LD 94 are used. The value 0 is also allowed, it means that the value defined in R2MFC table for incoming non-Tie trunk has to be used.

CAC conversion tables must be defined by the craftsperson in LD 15. Up to 32 tables can be configured. Then a configured conversion table entry can be assigned on each incoming DTI2 CIS route. For this, new prompt CAC_CONV is added to the CIS incoming route configuration. If no CAC conversion table is configured on the CIS route, the default table entry number 0 will apply on CIS incoming routes.

Default table entry is number 0 and is configurable. During upgrades, this table is defined in memory as shown in Table 37. As all values of default tables are set to 0, the R2MFC table will be used until default table number 0 is configured.

Table 37
CAC Conversion table entry 0 for CIS incoming DTI2 route (Part 1 of 2)

CIS CAC	MFC CAC	Description and CIS CAC meaning
CAC0	0	MFC value corresponding to CIS CAC 0 "Reserved"
CAC1	0	MFC value corresponding to CIS CAC 1 "Residential or business subset with the access to automatic regional, toll and international network"
CAC2	0	MFC value corresponding to CIS CAC 2 "Hotel subset with the access to automatic regional, toll international network"
CAC3	0	MFC value corresponding to CIS CAC 3 "Residential, business or Hotel subset with the access to local network only"
CAC4	0	MFC value corresponding to CIS CAC 4 "Business subset with the access to regional, toll, international network and to special service numbers; preferential access to regional and toll network"
CAC5	0	MFC value corresponding to CIS CAC 5 "Business subset of telecommunications administration with the access to automatic regional, toll, international network and to special service numbers free of charge"

Table 37
CAC Conversion table entry 0 for CIS incoming DTI2 route (Part 2 of 2)

CIS CAC	MFC CAC	Description and CIS CAC meaning
CAC6	0	MFC value corresponding to CIS CAC 6 "Toll Coin box and public call paystation with the access to automatic regional and toll network also the general purpose coin box with the access to local and toll network (paying in cash) and the coin box with access to special services only"
CAC7	0	MFC value corresponding to CIS CAC 7 "Business or residential subset with automatic access to regional, toll and international network plus to chargeable service numbers"
CAC8	0	MFC value corresponding to CIS CAC 8 "Business subset for data, facsimile and electronic mail with automatic access to regional, toll and international network."
CAC9	0	MFC value corresponding to CIS CAC 9 "Local call coin box"
DFLT	0	MFC value corresponding to CIS CAC 0 "Reserved"

Mapping CLID to ANI and OLI to ANI

The feature implements mapping of CLID on Euro-ISDN and enhances it on other ISDN interfaces.

The ANIE prompt in LD 16 configures an ANI table entry for the trunk route.

Building ANI for interfaces without calling number

The title of this subsection refers to routes whose connectivity does not give the possibility to receive a calling number (DTI2, Analog,...).

This case is similar to the situation where an Incoming route with calling number does not receive the calling number.

Mapping ANI to ANI in gateway CIS DTI2 to CIS

The feature CIS ANI Reception is developed concurrently to this one for incoming CIS DTI2 routes (see CIS ANI Reception in this document). For the gateway DTI2 CIS to CIS, on the outgoing CIS route the ANI is built using the ANI received on incoming route. The received ANI can be manipulated using ANI table entry configured on incoming CIS route.

The CAC used on outgoing route is the one received in incoming ANI. This CAC is copied into outgoing ANI without any modifications. If no ANI has been received on incoming CIS route. The CAC used is the one configured against CAC_CIS prompt on incoming CIS route.

Examples of Gateways Enhancements

The following pages contain examples of how to implement gateways enhancements.

Table 38
CAC conversion table, entry 1 content

MFC CAC	CIS CAC
CAC0	9
CAC1	8
CAC2	7
CAC3	6
CAC4	5
CAC5	4
CAC6	3
CAC7	2
CAC8	1
CAC9	0
Default	3

Example 1 - Gateway call through node 3 to CO C

In this example, Received CNI on route 320 is 1234, with CAC translated to DGT5. Outgoing ANI is 555 1 7476 4.

Table 39
Example 1 - Detail of built CAC

LEC Outgoing	Additional Digit	ANI DN Incoming	Category Code
555	1	7476	4

Example 2 - Gateway call through node 2 to CO B

In this example, received CNI on route 230 is 647678, with CAC translated to DGT 8. Outgoing ANI is 555 123 7678 1.

Table 40
Example 2 - Detail of built CAC

LEC Incoming	Additional digit	DNLG*CNI	Category Code
555	123	7678	1

Example 3 - Gateway call through node 1 to CO A

In this example, received CLID on route 120 is 25. Outgoing ANI is 444 123456123 25 5.

Table 41
Example 3 - Detail of built CAC

LEC Incoming	Additional digit	DNLG*CLID	Category Code
444	123456123	25	5

Operating parameters

This section lists the operating parameters for CIS ANI Digits Manipulation and CIS Gateways Enhancements.

CIS ANI Digits Manipulation operating parameters

The data in ANI is built only once at the beginning of the call. The data is not changed or re-downloaded for any kind of operation during a call. Therefore, if the call goes through any type of modification such as a transfer or call forward for instance, the ANI information sent when requested is of the original originator of the call.

This feature does not allow the user to associate ANI entry to attendant consoles. When an outgoing call is originated by an attendant, the ANI message is built using the old mechanism.

The ANI size flexibility is not supported by the analog trunks (E3W, X3W). For these type of trunks, the prompt ANSZ (LD 16) must be answered with seven. CIS standards recommend setting ANSZ to seven for CIS DTI2 ANSZ prompt.

CIS Gateways Enhancements operating parameters

ANI is downloaded to the card only once and immediately after trunk seizure. Thus calling number can be tandemed into ANI only if received before trunk is seized. That means that the calling number must be received at the same time that the called number or before enough digits of the called number are received to seize the CIS trunk. In case of gateway ISDN interface/CIS calling number (CLID) is always received before or at the same time as the called number. In case of gateway R2MFC/CIS calling number (CNI) can be received after trunk has been seized. Then incoming MFC route must be configured to request CNI before enough digits of called number are received to seize CIS trunk.

There is no cross checking of the ANI R_ENTRY associated with the route against the ANI R_ENTRY in the customer Data Block. If an ANI R_ENTRY is removed from the ANI table and the ANI R_ENTRY is stilled assigned to a route, ANI in this case is built using the default R_ENTRY, R_ENTRY number 0 and ERR9044 message is issued at call processing time.

While default R_ENTRY number 0 has not been configured, all its fields are empty except ADDG whose default value is 8. Thus the ANI built using this table is a repetition of 8. It is strongly recommended to configure this default table as soon as ANIC prompt is set to yes on an outgoing CIS route.

It is possible to associate an undefined ANI R_ENTRY to a route. Then, when constructing ANI if R_ENTRY is still undefined, default R_ENTRY number zero is used and ERR9044 message is issued at call processing time.

Due to the independence between ANI table and outgoing CIS route, it is possible to define a LEC, or ANDN greater than ANI size configured on outgoing route. Then most significant digits will be truncated. The way it is done is similar in case of call originated from a set or a trunk.

There is no cross checking of the CAC Conversion table entries associated with a CIS or R2MFC route against the CAC Conversion table in the customer Data Block. If a CAC Conversion table entry is removed from the CAC Conversion table and this entry is still assigned to a route, CAC in this case is built using the default CAC Conversion table entry, CIS_ENT number 0 for an incoming CIS route and MFC_ENT number 0 for an incoming MFC route. Additionally ERR9051 message is issued at call processing time.

It is possible to associate an undefined CIS_ENT or MFC_ENT to a route. Then, when constructing CAC if CIS_ENT or MFC_ENT is still undefined, default CIS_ENT or MFC_ENT (number 0) is used and ERR9051 message is issued at call processing time.

Feature interactions

This section identifies feature interactions for CIS ANI Digits Manipulation and Gateways Enhancements.

CIS ANI Digits Manipulation feature interactions

CIS Three Wire Analog Trunk

CIS ANI digits manipulation feature modifies Automatic Number Identification.

CIS Digital Trunk Interface

CIS ANI digits manipulation feature modifies Automatic Number Identification.

CIS MF Shuttle

CIS ANI digits manipulation feature modifies Automatic Number Identification.

CIS Gateways Enhancements

Some changes introduced in CIS ANI digits manipulation feature will impact on CIS Gateways Enhancements feature. These changes are listed below:

- The size of the ANI is configurable on a per outgoing CIS route basis.
- Valid responses for LEC, ADDG and ANDN programmed on the outgoing CIS route are modified.
- The additional digit(s) is(are) inserted between LEC and ANI DN when necessary to complete the ANI DN.
- Prompt ANIC (ANI Composing) added in LD 16 is also used in CIS Gateways Enhancements.

CIS Gateways Enhancements feature interactions

CIS Three Wire Trunk Analog

This development interacts with the feature CIS Three Wire Trunk Analog by changing how ANI is built in case of gateway from ISDN/DPNSS/R2MFC to CIS Analog.

CIS Digital Trunk Interface

The CIS Gateways Enhancements feature interacts with CIS Digital Trunk Interface by changing how ANI is built in case of gateway from ISDN/ DPNSS/R2MFC to CIS Digital.

CIS ANI Digits manipulation interaction

These interactions are already described in CIS ANI Digits Manipulation feature interactions.

Feature packaging

No new packages are introduced for these features.

The existing Commonwealth of Independent States Trunks (CIST) package 221 is required for these features to be operable.

CIS Multi Frequency Shuttle package is required if MF Shuttle signaling is to be used but is not mandatory for the CIS ANI digits manipulation and Gateways Enhancements functions.

The CIS DTI2 feature (including MF Shuttle) requires:

- Flexible Tones and Cadences (FTC) package 125
- International Supplementary Features (SUPP) package 131
- 2 Mbit Digital Trunk Interface (DTI2) package 129
- Flexible Numbering Plan (FNP) package 160

CIS Analog Three Wire Trunks feature requires:

- International Supplementary Features (SUPP) package 131
- Flexible Numbering Plan (FNP) package 160
- Flexible Tones and Cadences (FTC) package 125
- Trunk Fail Monitor (TFM) package 182
- IPE package (XPE) package 203 (required for outgoing X3W packs)
- Fast Tone and Digit Switch (FAST_TDS) package 87 (required for outgoing E3W package)

Feature implementation

CIS ANI Digits Manipulation feature

This section contains the overlay procedures required to configure the CIS ANI Digits Manipulation feature. The CIS Gateways Enhancements feature implementation section begins on page 1045.

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 - Configure ANI Table for telephones

2 LD 16 - Create or modify outgoing CIS route data block.

3 LD 10 - Define ANI entry for analog (500/2500) sets.

4 LD 10 - Configure the CAC for CIS and MFC signaling for analog 500/2500 sets.

5 LD 11 - Configure ANI entry for Meridian 1 proprietary sets.

6 LD 11 - Configure the CAC for CIS and MFC signaling for digital sets.

7 LD 27 - Configure ANI entry for Basic Rate Interface (BRI) lines.

8 LD 27 - Configure the CAC for CIS and MFC signaling for BRI sets.

9 LD 15 - Configure the ANI route table and CAC conversion table.

10 LD 16 - Configure the ANI table entry in the Route Data Block for incoming R2MFC route.

11 LD 16 - Configure the ANI table entry and CAC conversion table for Incoming CIS DTI2 route.

12 LD 16 - Configure the ANI table entry for all other route types (ISDN, incoming CIS analog, incoming DTI2 and DPNSS).

13 LD 88 - Configure the CAC for CIS signaling.

LD 15 - Configure ANI Table for telephones

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	ANI	Automatic Number Identification
...	...	...
ANLD	xx...xx	ANI listed directory number (for North American trunk)
CIS_ANI	YES (NO)	CIS ANI option. If CIS_ANI=YES, the next subprompts will be prompted, allowing the configuration of ANI entries for CIS ANI message composing. If CIS_ANI=NO, the following sequence of subprompts is skipped.
- S_SIZE	(0)-2000	Maximum number of ANI entries for sets that can be configured. If S_SIZE=0, next prompts are skipped. If <CR> is entered when REQ=NEW, it defaults to 0 and next prompts are skipped. The S_SIZE can't be decreased if the entries between the old size and the new one are not empty.
- S_ENTRY	aa Xaa Xaa Xbb	ANI entry for a set to be created or modified. ANI entry for a set to be deleted. ANI entries (aa-bb) for a set to be deleted. Prompted only if S_SIZE is greater than 0. ANI entries must be between 1 and S_SIZE. If REQ=NEW, this prompt and its subprompts are given only once. If REQ=CHG, they are repeated until S_ENTRY is answered with <CR>.
- - DNLG	0-(4)-15	Directory Number Length.

- - LEC	0-99...99 X	Local Exchange Carrier. Delete LEC.
- - ADDG	0-(8)-99...99	Additional digits, 1 to 15 digits long.
- - ANDN	0-99...99 X	Used as ANI DN if DN Length = 0. Delete ANDN.
...	...	...

LD 16 - Create or modify outgoing CIS route data block.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block
TKTP	DID COT	Direct Inward Dial. Central Office Trunk.
ICOG	OGT	Outgoing Route.
...	...	...
CCBA	(NO) YES	Deny/Allow collect call blocking.
CISR	YES (NO)	CIS Route
- ANSZ	(7)-15	ANI information size. Response is length of LEC+ANI DN. For analog routes, the only valid response is seven (7).

- ANIC	(NO) YES	ANI Composing Prompted for outgoing CIS route. If ANIC=NO, old ANI composing is used: if the originator of the call is a set, ANI message will consist of the CAC of the originator + the LEC of the outgoing route + the DN of the originator or the ANDN of the outgoing CIS route, depending on the class of service (DNAA/DNAD) of the set. If the originator of the call is an incoming route, the components of the ANI message are retrieved from default ANI entries and/or from the data block of the outgoing CIS route (See "CIS Gateways Enhancements feature implementation" on page 1045.) If the outgoing CIS route requires new ANI composing to be used (prompt ANIC=YES), the following is done: If no entry is associated to the calling set (ANIE=0), then old ANI composing is used. If an ANI entry is associated to the calling set (ANIE has a non-zero value), but the associated ANI entry is not configured, then old ANI composing is used. If an ANI entry is associated to the calling set (ANIE has a non-zero value), and the associated ANI entry is configured, then ANI table will be used for building the ANI message: none of the components of the ANI message will be retrieved from the data block of the outgoing CIS route.
- LEC	0-99...99	Local Exchange Code of the route. It can be from 0 digit long up to ANSZ digit long. Prompted for outgoing CIS route. Used for building ANI message if ANIC is NO or if ANIC=YES but the ANI entry associated with the originator of the call is not configured.
	X	Remove LEC.

- ADDG	0-(8)-99...99	Additional digits used in ANI message if ANIC is NO or if ANIC=YES but the ANI entry associated with the originator of the call is not configured. It is used to complete ANI DN if LEC+ANI DN consists of less than ANSZ digits. Prompted for outgoing CIS route. It can be from 1 digit long up to ANSZ digit long.
- ANDN	0-99...99	Default ANI DN. It can be from 0 digit long up to ANSZ digit long. Prompted for outgoing CIS route. Used for building ANI message if ANIC=NO and DN of set is not allowed to be sent (CLS DNAD). Also used if ANIC=YES but the ANI entry associated with the originator of the call is not configured, and DN of set is not allowed to be sent (CLS DNAD).
	X	Remove ANDN.

LD 10 - Define ANI entry for analog (500/2500) sets.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	500	500 set.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...	...	...
DN	xxxx ccc	Directory Number, CLID entry.
- MARP	(NO) YES	Deny/Allow Multi Appearance Redirection Prime.
- CPND	aaaa	Calling Party Name Display.
- VMB	aaaa	Voice Mailbox.

- ANIE	(0)-N	ANI Entry: it is of (0)-N where N=S_SIZE in customer data block. If ANIE=0, no entry is associated with the set. The old mechanism will be used for building the ANI message. If ANIE is of 1-N: • If ANIC = YES for the outgoing CIS route where the call takes place, then the components of the ANI message are retrieved from the ANI entry in Customer Data Block, if configured. • If the given ANI Entry is not configured, or if ANIC = NO for the outgoing CIS route where the call takes place, then the old mechanism is used for building the ANI message.
...	...	...

LD 10 - Configure the CAC for CIS and MFC signaling for analog 500/2500 sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Type of data block for analog (500/2500) set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...	...	
SFLT	a..a	Secretarial Filtering.
CAC_CIS	0-(3)-9	CIS ANI Category Code.
CAC_MFC	(0)-10	MFC CNI Category Code.

LD 11 - Configure ANI entry for Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	xxxx	Meridian 1 proprietary set.
TN		Terminal Number
	l s c u	For Large Systems
	c u	For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...	...	...
KEY	xx AAA yyyy... ccc	xx = key number.
		AAA = Key Function.
		yyyy = DN or other data related to key function.
		ccc = CLID entry.
- MARP	(NO) YES	Deny/Allow Multi Appearance Redirection Prime.
- CPND	aaaa	Calling Party Name Display.

- VMB	aaaa	Voice Mailbox.
- ANIE	(0)-N	ANI Entry: It is of (0)-N where N=S_SIZE in customer data block. If ANIE=0, no entry is associated with the DN key. The old mechanism will be used for building the ANI message. If ANIE is of 1-N:. • If ANIC=YES for the outgoing CIS route where the call takes place, then the components of the ANI message are retrieved from the ANI entry in Customer Data Block, if configured. • If the given ANI Entry is not configured, or if ANIC=NO for the outgoing CIS route where the call takes place, then the old mechanism is used for building the ANI message.

LD 11 - Configure the CAC for CIS and MFC signaling for digital sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaa	Meridian 1 proprietary set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...	...	
CAC_CIS	0-(3)-9	CIS ANI Category Code.
CAC_MFC	(0)-10	MFC CNI Category Code.

LD 27 - Configure ANI entry for Basic Rate Interface (BRI) lines.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Administer Terminal Service Profile on Digital Subscriber Loop.
...	...	...
SPID	xxxx	Service Profile ID.
DN	xxxx ccc	Directory Number (DN) and Calling Line Identification (CLID) entry. xxxx = DN ccc = CLID
- CT	aaaa	Call Types for DN aaaa = VCE or DTA.
- MCAL	1-(4)-8	Maximum Calls allowed per DN
- CLIP	(YES) NO	Allow/Deny Calling Line Identification Presentation for incoming calls.

- PRES	(YES) NO	Allow/Deny Presentation of CLID to far end on outgoing calls.
- ANIE	(0)-N	ANI entry. It is of (0)-N where N=S_SIZE in customer data block. If ANIE=0, no entry is associated with the DN key. The old mechanism is used for building the ANI message. If ANIE is 1-N: • If ANIC=YES for the outgoing CIS route where the call takes place, then the components of the ANI message are retrieved from the ANI entry in Customer Data Block, if configured. • If the given ANI Entry is not configured, or if ANIC=NO for the outgoing CIS route where the call takes place, then the old mechanism is used for building the ANI message.

LD 27 - Configure the CAC for CIS and MFC signaling for BRI sets.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DSL	Digital Subscriber Loop.
DSL	 l s c dsl c u dsl	Digital Subscriber Loop For Large Systems For Small Systems and Succession 1000 systems
...	...	
CAC_CIS	0-(3)-9	CIS ANI category code.
CAC_MFC	(0)-10	MFC CNI category code.

CIS Gateways Enhancements feature implementation

This section contains the overlay procedures required to configure the CIS Gateways Enhancements feature.

Feature implementation for R2MFC route

In case of gateway R2MFC/CIS it must be noticed that to be tandemed CNI must be received on R2MFC route before outgoing CIS trunk is seized. This is possible using an up-front CNI request. Up-front CNI request is defined on R2MFC route by giving a non zero value to prompt NCNI. Configure the ANI table entry and CAC conversion table for Incoming CIS DTI2 route using LD 16.

LD 15 - Configure the ANI route table and CAC conversion table.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data. REQ = NEW, default table R_SIZE (0) is created. REQ = NEW, default table CACC(0) is created. REQ = NEW, R_SIZE defaults to number of ANI entries. for incoming route created during conversion.
TYPE:	ANI	Automatic Number Identification
....	...	....
ANLD	xx...x	ANI Listed Directory Number. Only included here for clarification if this prompt appears. Only applies to North American ANI.
CIS_ANI	(NO) YES	Deny/Allow configuration of ANI entries for CIS ANI message.
...	...	
- R_ SIZE	(1)-512	Maximum number of ANI entries that can be configured for incoming routes. This maximum number is limited to 512 as it is the maximum number of routes. Entries must be empty to decrease R_SIZE.

- R_ENTRY	aa	ANI entry for an incoming route to be created or modified.
	Xaa	ANI entry for an incoming route to be deleted.
	Xaa Xbb	ANI entries for an incoming route between aa and bb to be deleted.
	<CR>	Exit.
		R_ENTRY is repeated until <CR> is entered. ANI entries must be between 0 and (R_SIZE-1). For REQ=NEW, only default table 0 is configurable. An R_ENTRY can be deleted even if still assigned on an incoming route.
- - DNLG	0-(4)-15	DN Length
		Number of digits of the Calling Line ID (CLID), Originating Line Identifier (OLI), Calling Number Identification (CNI) to use in the ANI message, starting with the less significant digits.
- - LEC	0-99..99	Local Exchange Code, 1 to 15 digits.
	X	Remove LEC.
- - ADDG	0-(8)-99...99	Additional digits, 1 to 15 digits. Used to complete ANI Directory Number (ANDN) if Local Exchange Carrier (LEC)+ANDN is less than ANSZ digits defined in LD 16.
- - ANDN	0-99...99	Used as ANI DN if calling number is not available or DNLG=0. Up to 15 digits may be entered.
	X	Remove ANDN.
CACC	(NO) YES	Deny/Allow Calling Party Category Code (CAC) Conversion table option.

-MFC_ENT		CAC conversion table to convert MFC CAC into CIS CAC for use on R2MFC routes.
	aa	CAC conversion table entry to be created or modified.
	Xaa	CAC conversion table entry to be deleted.
	Xaa Xbb	CAC conversion table entries between aa and bb to be deleted.
	<CR>	Exit MFC CAC conversion table, gives CIS_ENT prompt. This prompt is repeated until <CR> is entered. It is prompted if CACC=YES. CAC conversion table entries must be between 0 and 31. For REQ=NEW, only default table 0 is configurable. An ENTRY can be deleted even if still assigned on an incoming route.
-- CAC0	0-(3)-9	CIS value corresponding to MFC DGT0
-- CAC1	0-(3)-9	CIS value corresponding to MFC DGT1
-- CAC2	0-(3)-9	CIS value corresponding to MFC DGT2
-- CAC3	0-(3)-9	CIS value corresponding to MFC DGT3
-- CAC4	0-(3)-9	CIS value corresponding to MFC DGT4
-- CAC5	0-(3)-9	CIS value corresponding to MFC DGT5
-- CAC6	0-(3)-9	CIS value corresponding to MFC DGT6
-- CAC7	0-(3)-9	CIS value corresponding to MFC DGT7
-- CAC8	0-(3)-9	CIS value corresponding to MFC DGT8
-- CAC9	0-(3)-9	CIS value corresponding to MFC DGT9
-- DFLT	0-(3)-9	CIS value used when MFC CAC has not been received, or MFC CAC received is not in the MFC CAC list of this table

LD 16 - Configure the ANI table entry in the Route Data Block for incoming R2MFC route.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	RDB	Route Data Block.
...	...	...
MFC	YES	Configure Multifrequency Compelled signaling.
- MFCI	x	MFC Incoming Table.
- MFCO	x	MFC Outgoing Table.
...	...	
ICOG	IAO	Incoming and Outgoing.
	INC	Incoming.
NCNI	1	Request CNI after the defined number of digits are received. CNI is requested during incoming MFC call after the first digit of dialed number has been received.
ANIE	(0)-x	ANI table Entry for Route (configured under prompt R_ENTRY, OVL 15). x = R_SIZE-1. R_Size is the maximum table entry number that can be configured. (R-SIZE is defined in LD 15).
CAC_CONV	(0)-31	CAC conversion table number for CIS Gateway. Configured against MFC_ENT in LD 15. Prompted only for non “outgoing only” R2MFC route.

LD 16 - Configure the ANI table entry and CAC conversion table for Incoming CIS DTI2 route.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	RDB	Route data block
...	...	...
DGTP	DTI2	2 Mbit Digital Trunk Interface.
...	...	
ICOG	INC	Incoming route.
...	...	...
ANIE	(0)-x	ANI table Entry for Route (configured under prompt R_ENTRY, OVL 15). x = R_SIZE-1. R_Size is the maximum table entry number that can be configured. (R-SIZE is defined in OVL 15). Default table entry is number 0.
CISR	YES	CIS Route
...	...	...
- CAC_CONV	(0)-31	CAC conversion table number for CIS-to-R2MFC gateway, configured against CIS_ENT in CDB. Prompted only for incoming CIS DTI2 route.
CAC_CIS	0-(3)-9	CIS ANI Category Code

LD 16 - Configure the ANI table entry for all other route types (ISDN, incoming CIS analog, incoming DTI2 and DPNSS).

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	RDB	Route Data Block.
...	...	
ISDN	YES	Enable ISDN
...	...	...
ICOG	IAO	Incoming and outgoing.
	ICT	Incoming.
...	...	
ANIE	(0)-x	ANI table Entry for Route (configured under prompt R_ENTRY, LD 15). x= R_SIZE-1. R_Size is the maximum table entry number that can be configured.
...	...	...
CAC_CIS	0-(3)-9	CIS ANI Category Code

LD 88 - Configure the CAC for CIS signaling.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AUB	Authcode Data Block
...	...	...
CLAS	(0)-115	Classcode value assigned to authcode (NAUT).
...	...	...
NCOS	(0)-99	Network Class of Service group number.
CAC_CIS	0-(3)-9	CIS ANI category code.

Feature operation

No specific operating procedures are required to use this feature.

CIS ANI Reception

Contents

This section contains information on the following topics:

Feature description 1053
Operating parameters 1061
Feature interactions 1061
Feature packaging 1061
Feature implementation 1062
Feature operation 1067

Feature description

The Commonwealth of Independent States (CIS) Automatic Number Identification (ANI) Reception feature allows the system to receive the Automatic Number Information from the CIS Central Office (CO) calling party on incoming local calls. The CIS Public Telephone Network does not provide ANI information on the incoming toll calls.

The CIS ANI Reception feature works on the CIS Digital Trunk Interface (DTI) feature. It requires CIS DTI2 card, vintage NTCG01AC (Large Systems) or NTCG02AC (Small Systems and Succession 1000 systems).

The ANI digits received from the CIS CO are used by the system as the R2MFC Calling Number Identification (CNI) digits. A list of uses for the ANI digits is found on page 1054. The ANI digits are also displayed on the display of the proprietary set or on the attendant console display.

ANI Reception is performed in one of two ways:

- ANI request is issued automatically by the incoming local CIS DTI2 trunk during the call setup.
- ANI request is issued by the incoming local CIS DTI2 trunk upon a manually solicited request from the proprietary set with display or from the attendant console. The request to receive the ANI information is invoked by pressing a calling number display key on the attendant console source key or on the proprietary set by pressing the display key followed by the trace key. Press trace key for active call for Proprietary sets and attendant consoles. The trace key can be used instead of the source key for Malicous Call Trace (MCT) for sets and consoles. An ANI request can also be made by 500/2500 sets by entering the Flexible Feature Code (FFC) assigned for MTRC in LD 57.

Automatic ANI request

The automatic ANI request is sent by the CDTI2 card to the CIS CO before the incoming local call is answered. If the incoming trunk operates in the decadic, or Dial Pulse (DP) mode, the ANI request is sent to the CIS CO after all dialed digits have been collected from the CIS CO (see Figure 15.) If the trunk operates in the MF Shuttle mode, the ANI request is sent after the end of the MF Shuttle dialing (see Figure 16 on page 1057). The ANI digits are uploaded to the system.

The Automatic ANI request option may be used only in conjunction with the DN Size Feature. The DN Size Flexible (using the SSL tables) or Fixed should be defined for the incoming CIS DTI2 DID route before setting the automatic ANI option to “Yes”.

The ANI digits are used in the following way:

- They are tandemed as the Calling Line Identification (CLID) Originating Line Identifier on DPNSS (OLI) to the Integrated Services Digital Network (ISDN)/Digital Private Network Signalling System (DPNSS) gateways, Basic Rate Interface (BRI) gateways
- They are mapped into the Multi-frequency Compelled Signaling complying with CCITT R2 specification (R2MFC) Calling Number Identification R2MFC (CNI)

- They are displayed on the display of the proprietary sets and on the attendant consoles
- They are stored in the Call Detail Recorder
- They are sent through the Meridian Link and the ICCM link using the fields dedicated for the R2MFC CNI digits

The translation of the dialed number which is received from the CIS CO is postponed until the CDTI2 card informs the system that the ANI digits have been received. If the ANI reception report does not arrive from the card, the call is treated after the ANI timer expires. The treatment for the call that failed to provide the automatic ANI is configured on the route data block level.

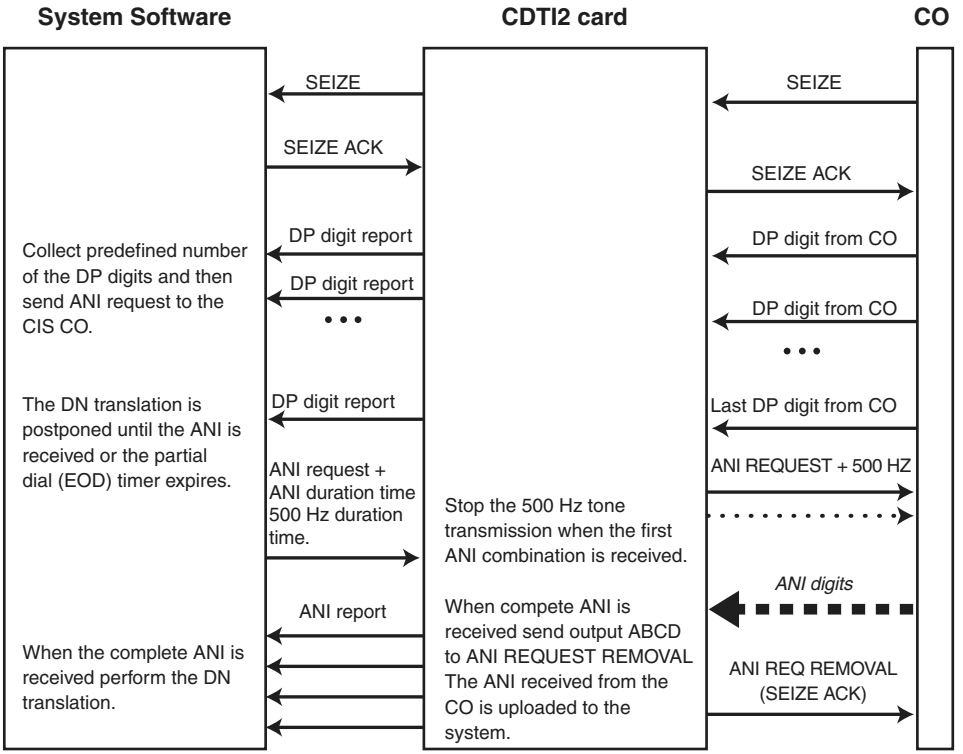
Solicited ANI request

The solicited ANI request is issued upon a manual request from the Meridian 1 proprietary set using the display key or from the attendant console using the trace key. The received ANI information is displayed on the display of the proprietary set or attendant console and it is also used for the CDR, call trace. If the request is performed using the Trace key, the MCT record is printed also on the MCT TTY. The system sends the ANI request message to the CDTI2 card when a user presses the DN key after the set has entered display mode using the Digit Display key or after the Trace key was pressed. The CDTI2 card performs the ANI interaction and uploads the received ANI to the system (see Figure 17).

The ANI is stored in the unprotected trunk data block. The solicited ANI request may be performed an unlimited number of times during a single call. Each time, the new ANI replaces the previously received ANI (if there is one).

When solicited ANI request is initiated using the Trace key, the Malicious Call Trace feature functionality remains unchanged, the ANI request is sent in addition to the normal MCT activities. The ANI information, when received from the card, is also printed on the MCT TTY as MCT ticket.

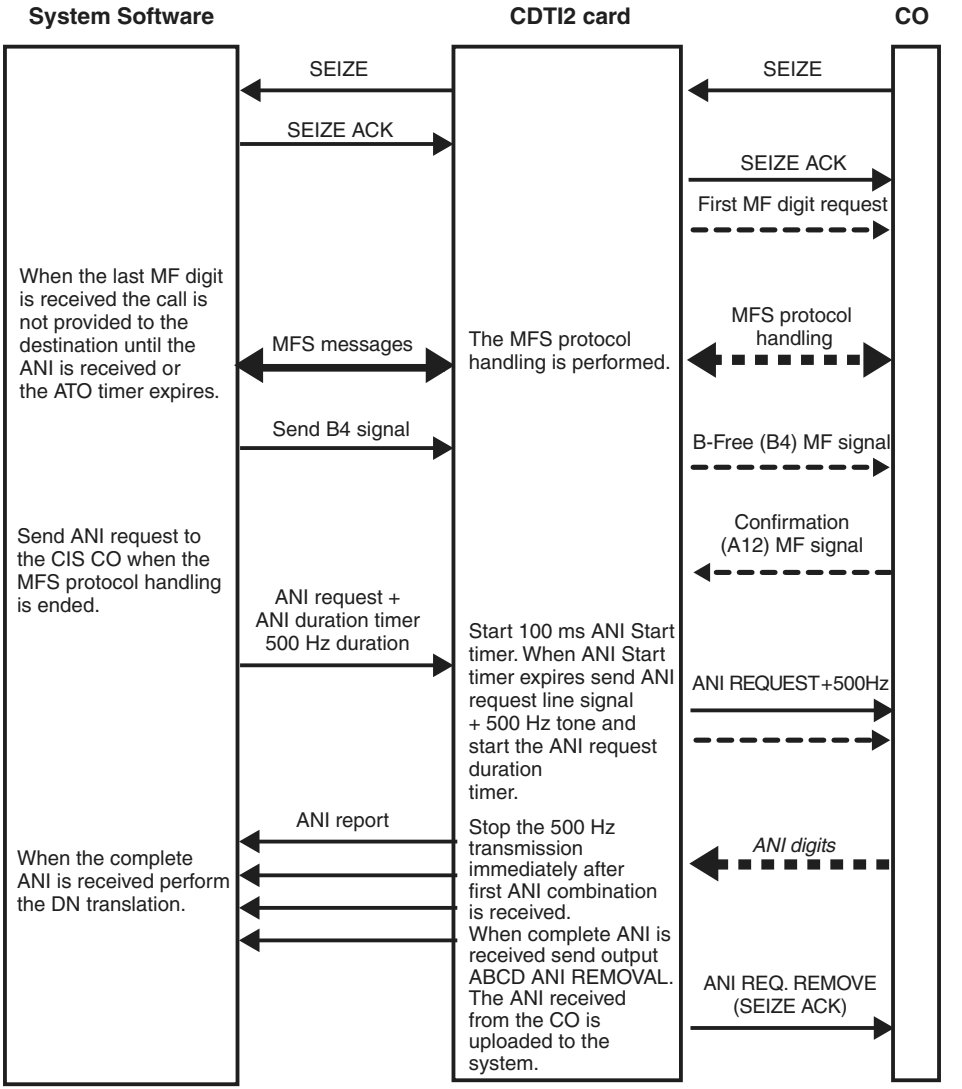
Figure 15
Automatic ANI request for incoming local call (decadic dial pulse mode)



Note:
From this point the call progress flow continues in the same way as it is done without the ANI reception.

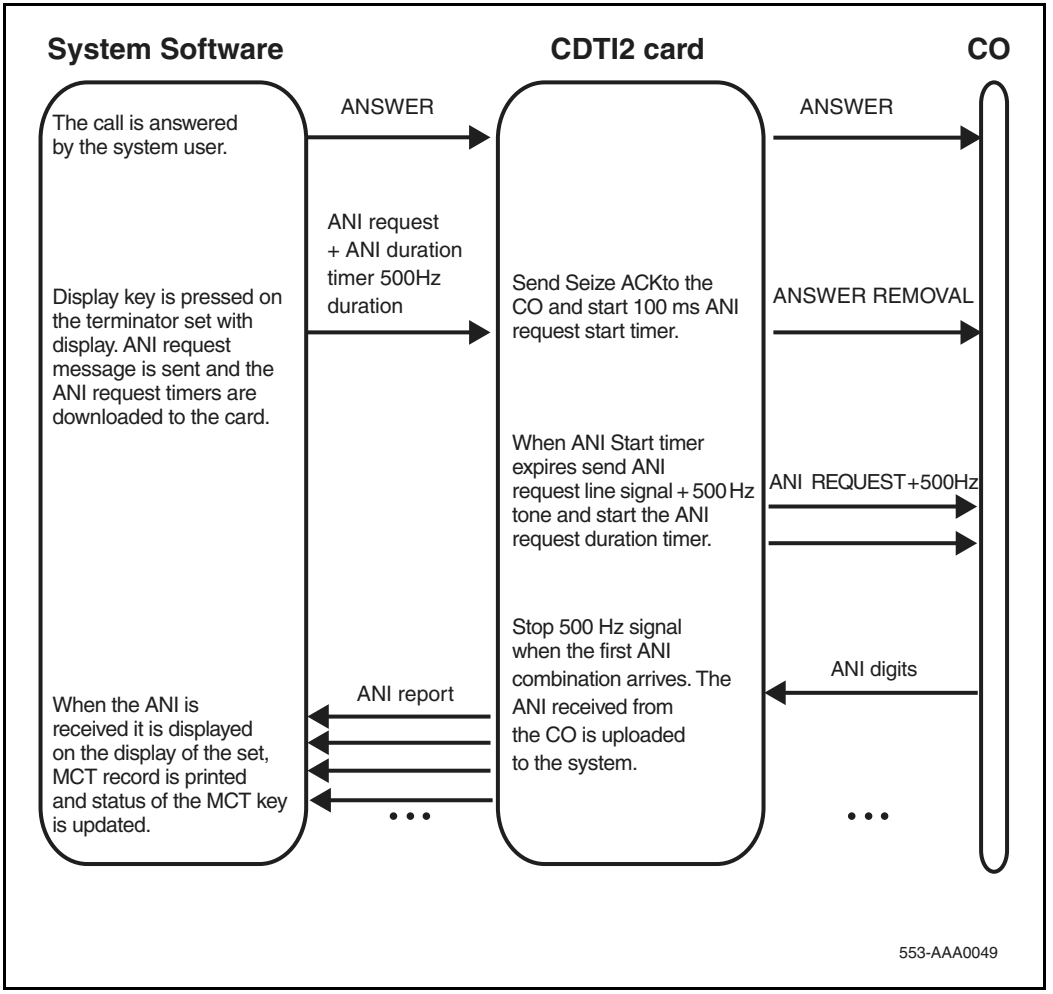
553-AAA0037

Figure 16
Automatic ANI request for the MF shuttle call



Note:
From this point the call progress flow continues in the same way as is done without the ANI reception.

Figure 17
Manually solicited ANI request



ANI Gateways

The ANI digits which are received from the CIS CO party as a response to the automatic ANI request are propagated to the system terminating party if it is capable of receiving the CNI digits.

The ANI digits are propagated to the following terminating types:

- R2MFC trunks - the ANI to R2MFC CNI mapping is performed in the following way: all the ANI digits except for the ANI Calling Party Category Code (CAC) are used for the CNI composition, the ANI CAC is converted to the Multi-frequency Compelled (MFC) CNI CAC according to the CAC conversion tables.
- ISDN trunks - Meridian Customer Defined Integrated Services Digital Network (MCDN), EuroISDN: European Integrated Services Digital Network (EURO), Q Signaling (QSIG), DPNSS - the ANI to CLID/OLI mapping is based on the R2MFC CNI to CLID mapping.
- CIS trunks - the ANI to ANI mapping is implemented in the framework of the CIS ANI Digits Manipulation and Gateways Enhancements feature described in this document. The ANI information that is received from the incoming CIS DTI2 trunks is used by the CIS Gateways Enhancements feature to compose the ANI information to be downloaded to the outgoing CIS trunks.

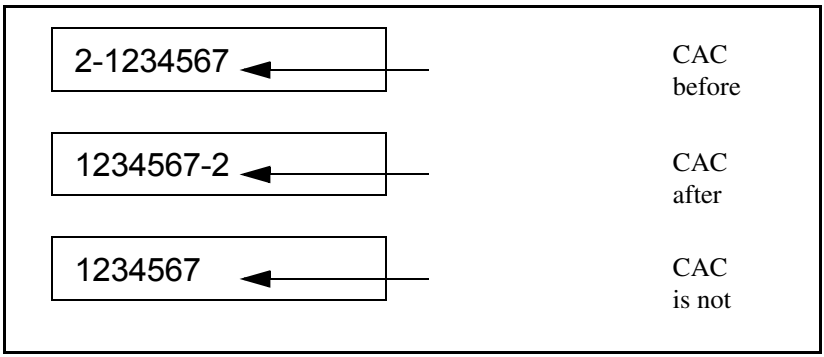
ANI Digits Display

The ANI digits are displayed on the display of the proprietary set or of the attendant console. The ANI digits are displayed starting from the left side of the display. The CAC display option is configured on the route data block level and it may be set to one of the following options (see Figure 18):

- display CAC before the ANI number
- display CAC after the ANI number
- do not display CAC

The CAC digit is separated from the ANI number by the minus sign. When displaying ANI, there are several options available for CAC display. If ANI request is repeated several times during a single call, each time new ANI digits overwrite old ANI digits on the display. If the DN key that requested the ANI information is placed on hold when the ANI digits arrive from the CDTI2 card, the ANI digits are not displayed. In this case, when the call is returned from hold, the ANI digits are displayed.

Figure 18
ANI Display options



ANI Digits in CDR

The ANI digits are placed in the CDR at the place intended for the R2MFC CNI digits. The CAC may be stored in the CDR together with the ANI digits. The presentation of the CAC in the CDR is configured at the route data block level as shown in the dash list below. The CAC may be either:

- placed before the ANI digits
- placed after the ANI digits
- not placed in the CDR

The CDR records containing ANI are generated only if the CDR options are configured in the customer data block (LD 15) as described in the CDR NTP.

The feature does not change the CDR output formats. The ANI digits are stored in the CDR in the same way as the R2MFC CNI digits.

Operating parameters

This feature requires the CIS DTI2 card NTCG01 vintage AC for Large Systems, and the CIS DTI2 card NTCG02 vintage AC for Small Systems.

Feature interactions

CIS Digital Trunk Interface and CIS Multifrequency Shuttle (MFS)

The feature is based on the CIS DTI2 interface features. The feature enhances the capabilities of CIS DTI2 but does not change previous functionality.

Malicous Call Trace (MCT)

The feature allows MCT to receive ANI digits on incoming local CIS DTI2 calls. There is no MCT feature that can be activated from the system on the CIS CO. When the ANI digits are received from the CO, they are printed on the MCT TTY and displayed on the display of the set or console that activated the MCT feature.

Feature packaging

No new package is introduced for this feature. The following packages are required:

- 2 Mbit Digital Trunk Interface (DTI2) package 129
- International Supplementary Features (SUPP) package 131
- Commonwealth of Independent States Trunks (CIST) package 221
- Optional - Malicous Call Trace (MCT) package 107

Feature implementation

This section contains the overlay procedures required to configure the ANI Reception feature.

> ***Note:*** If Malicous Call Trace is used, turn to the Software Features Guide for detailed MCT implementation instructions.

- **LD 73** - The response MFA - Multifrequency Advanced has been added to the CISFW prompt to support the ANI Reception and the Firmware Dial Tone Detection features on the NTCK01AC and NTCK02AC cards. When operating in the CIS mode, the NTCG01AA/02AA cards should be configured as DP, and the NTCG01AB/02AB cards should be configured as MFS.
- **LD 14** - The CNA may be defined for the incoming DID DTI2 trunk if it is configured on the CDTI2 loop with the CIS Firmware version set to MultiFrequency Advanced in LD 73.
- **LD 16** -Set the prompt CISR to YES to allow access to the CIS Route prompts.

The following options are added for the CIS DTI2 routes:

- The Automatic ANI (AANI) option determines if the automatic ANI request should be sent to the CIS CO party at the end of dialing on incoming local calls. If the ANI information is required for gateway calls this option should be set to YES.
- The ANI Failure Treatment (ANFT) option defines the treatment for incoming calls which failed to provide ANI (see Note). The possible options are:
  - to provide the call to the required destination using the alternative ANI to indicate the ANI Reception failure
  - to drop the call
  - to transfer the call to the predefined intercept DN using the Alternative ANI to indicate the ANI Reception failure
- The Intercept DN (ITDN) prompt defines the intercept DN if the intercept treatment option is selected.

Note: The Alternative ANI is composed from the access code of the incoming CIS DTI2 DID route and the number of the incoming trunk within the route.

- The ANI TimeOut (ATO) timer which was used only for Outgoing CIS trunks is now also used for incoming CIS trunks to define timeout for the automatic ANI digits reception. The termination of the incoming call is delayed until either the ANI digits are received from the CDTI2 card, the ANI failure report is received from the CDTI2 card or the ATO expires. The ATO timer should be set at least twice as large as the following ARD timer.
- ANI Request Duration (ARD) timer is added to the route timers, the timer defines how long the CDTI2 card waits for the ANI information after sending the ANI REQUEST line signal. When the ARD timer expires for the first time the CDTI2 card performs an additional attempt to request the ANI information from the CIS CO. If the timer expires after the second attempt the ANI reception failure is reported to the system. The ARD timer is downloaded to the CDTI2 firmware. The timer should not exceed half of the ATO timer.
- CAC Display (CACD) option defines how the Category Access Code (CAC) is displayed on the proprietary set or attendant console. The possible options are:
  - to display CAC before the ANI
  - to display CAC after the ANI or
  - not to display CAC
- CAC in CDR (CACC) option defines how the Category Access Code (CAC) is stored in the CDR. The possible options are to:
  - store CAC before the ANI
  - store CAC after the ANI
  - not store CAC.

Task summary list

The following is a summary of the tasks in this section:

1 LD 73 - Define Multifrequency Advanced (MFA) as firmware type definition.

2 LD 14 - Add Calling Number Identification Allowed (CNA) as a class of service for incoming Direct Inward Dial (DID) CIS DTI2 trunks.

3 LD 16 - Define the CIS Route.

LD 73 - Define Multifrequency Advanced (MFA) as firmware type definition.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	DTI2	Digital Trunk Interface loops.
FEAT	LPTI	Loop Timers and additional definitions.
CDTI2	YES	CDTI2/CSDTI2 card
...	...	
CISFW	MFA	Multifrequency Shuttle protocol handling + ANI Reception + Firmware Dial Tone Detection capabilities are supported by the card.
- MFSL	(0)-3	The MFS signals transmission level. The prompt appears when CISFW is set to MFS or MFA and the CISMFS package is equipped. The transmission level may be set to the following values: 0 = -7.3 dB 1 = -5.0 dB 2 = -3.5 dB 3 = 0 dB
- 500L	(0)-1	ANI request tone (500 Hz) transmission level. The prompt appears when CISFW is set to MFA. The transmission level may be set to the following values: 0 = -7.3 dB 1 = -3.5 dB

LD 14 - Add Calling Number Identification Allowed (CNA) as a class of service for incoming Direct Inward Dial (DID) CIS DTI2 trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DID	Direct Inward Dial trunks.
...	...	...
CLS		Class of Service.
	CNA	Allow Calling Number Identification for incoming CIS DTI2 DID trunks.

LD 16 - Define the CIS Route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
TKTP	DID	Direct Inward Dial.
...	...	...
DTRK	YES	Digital Trunk Route
- DGTP	DTI2	Digital Trunk Type
...	...	...
ICOG	ICT	Incoming Trunk.
...	...	...
CNTL	YES	Allows change to controls or timers.
TIMR	ARD 512-(1024)-2048	ANI Request Duration timer, defines duration of the ANI request signal. The timer is stored as increments of the 256 ms.

TIMR	ATO 2048-(2560)-5120	ANI Timeout timer. Defines how long the system waits for the ANI information from the CIS CO. If timer expires before the ANI is uploaded from the CDTI2 card, the system treats the call as it is defined in the ANI Failure Treatment option. The ATO should be at least twice as large as the ARD.
CISR	YES	CIS Route
- CACD		The option defines how the CAC is displayed on the display of the set or console. The option also controls presentation of the CAC in the messages to the auxiliary processors. The options are:
	(NO) BEF AFT	NO = do not display CAC. BEF= display CAC before ANI. AFT= display CAC after ANI.
- CACC		Defines how CAC is stored in CDR.
	(NO)	NO = do not store CAC
	BEF	BEF = store CAC before ANI
	AFT	AFT = store CAC after ANI
- AANI		The option defines if the Automatic ANI request should be sent to the CIS CO when the incoming calls are originated from the CIS CO to the trunks within this route.
	(NO)	NO = do not send automatic request.
	YES	YES = send automatic request.
- ANFT		The prompt defines the ANI Failure Treatment option. It is prompted only if the AANI is set to YES. The possible options are:
	(CONT)	Provide call to the required destination.
	FAIL	Drop call.
	ITDN	Transfer call to intercept DN.
- - ITDN	<DN>	Intercept DN (up to 8 digits) defines the DN to transfer the incoming calls which failed to provide ANI. The prompt appears if the ANFT option is set to ITDN.

Feature operation

Manual ANI request is made by pressing a calling number display key on the Attendant Console or on the proprietary set followed by pressing the SCR key or by pressing the TRC key on the proprietary set or Attendant Console.

CIS Toll Dial Tone Detection

Contents

This section contains information on the following topics:

Feature description 1069
Operating parameters 1073
Feature interactions 1073
Feature packaging 1073
Feature implementation 1073
Feature operation 1075

Feature description

The Commonwealth of Independent states (CIS) Toll Dial Tone Detection feature allows the system to detect a dial tone from a CIS Toll Central Office (CO) on outgoing toll calls. When received, the tone indicates that the CIS CO is prepared to collect dial pulse (decadic) digits from system for outgoing toll calls. The feature is implemented only for CIS DTI2 trunks.

The CIS DTD feature introduces CIS toll outpulsing criteria, which are terms that define conditions that need to be satisfied to allow the system to start the outpulsing of the decadic digits on the outgoing Toll CIS DTI2 calls (see Figure 19).

The criteria is composed as a combination of two events: dial tone detection and ANI interaction. It may include only dial tone detection, only ANI Interaction, dial tone or ANI, dial tone and ANI. The criteria is defined at the route data block level. The detection is performed by the CIS DTI2 cards (NTCG01AC and NTCG02 AC) and when the criteria is satisfied the system receives a report from the card. The system postpones the outpulsing of the digits until the report. If the report does not arrive before the ATO timer expiration, defined in LD 16, the outpulsing may continue, or the call may be disconnected and busy tone provided to customer.

Outgoing toll CIS DTI2 calls can be made using the indirect or direct connection method (see Figure 20 and Figure 21).

Figure 19
CIS network block diagram

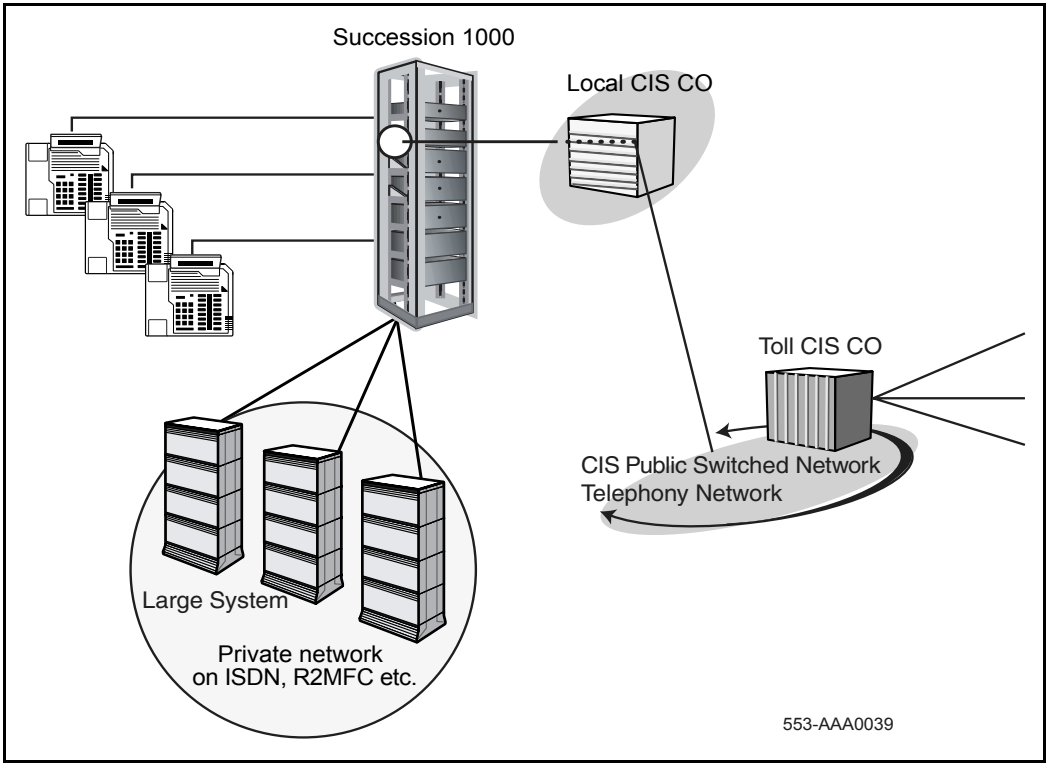


Figure 20
Indirect outgoing toll call with dial tone detection

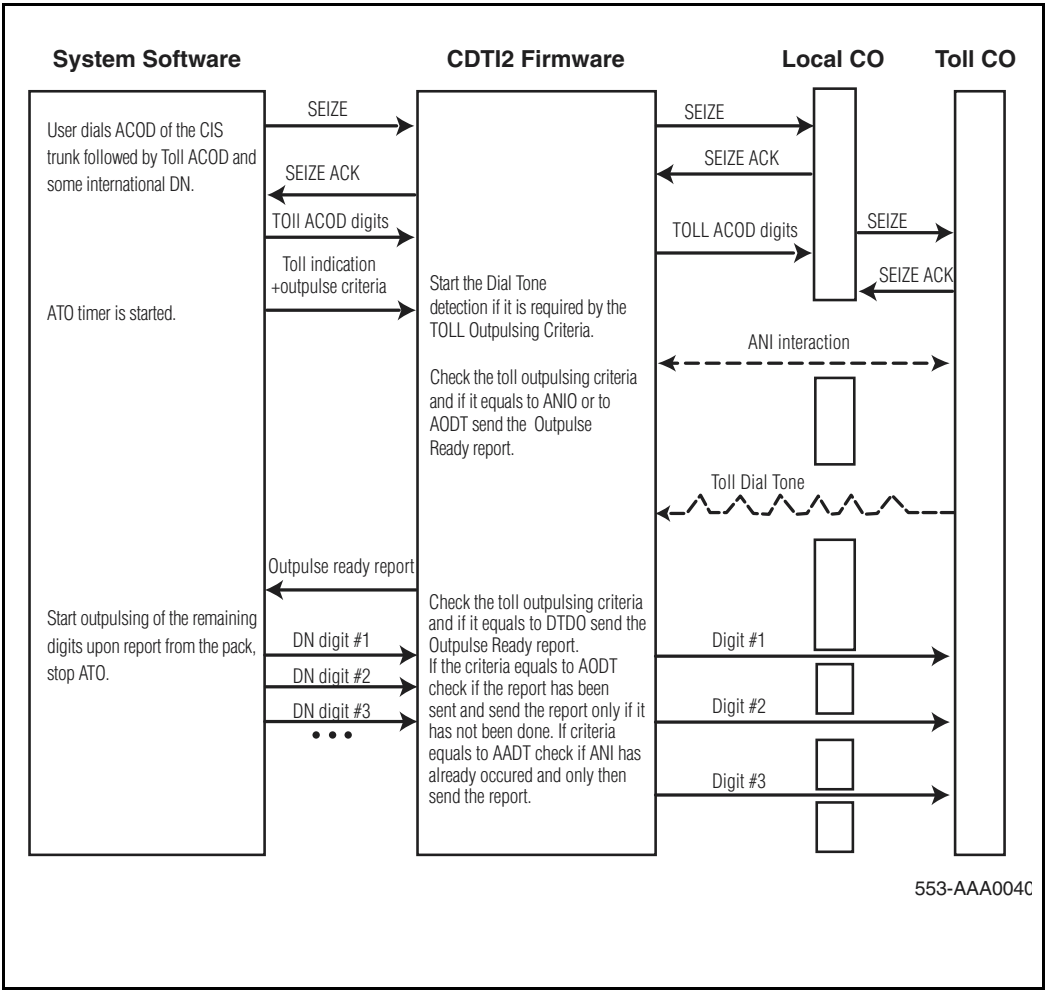
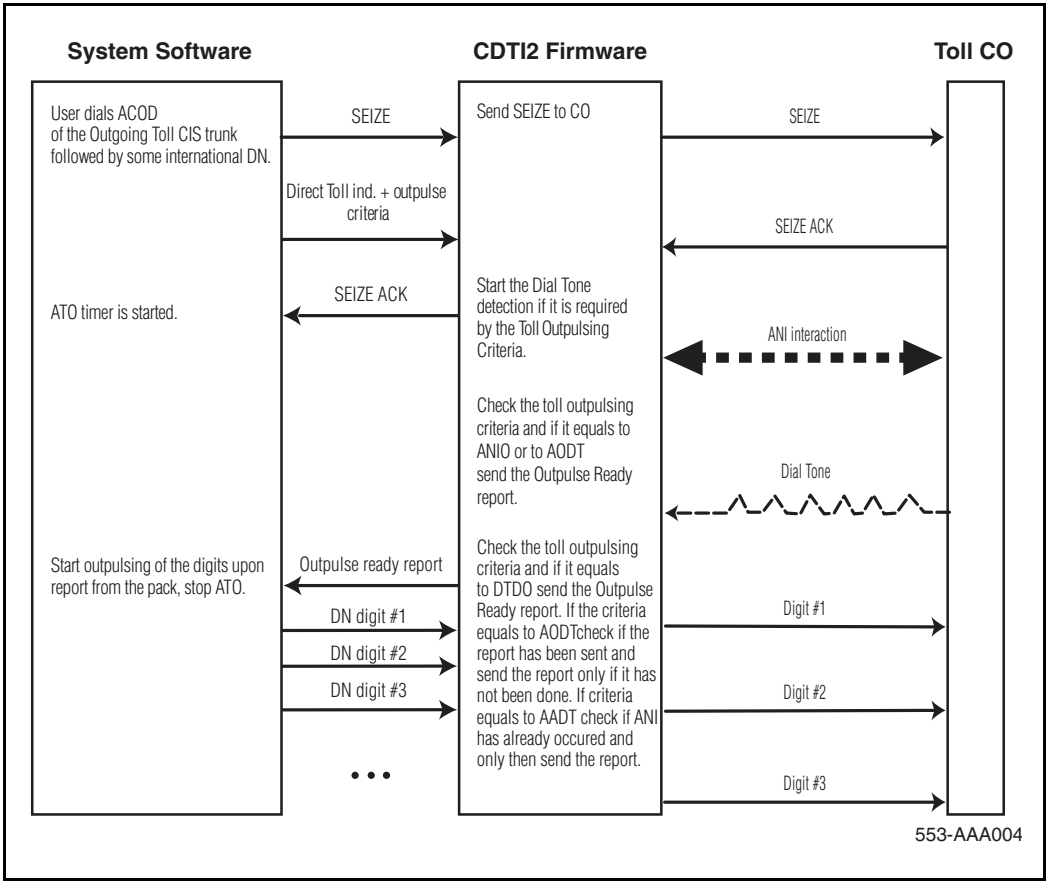


Figure 21
Direct toll call with dial tone detection



Operating parameters

This feature requires the CIS DTI2 card NTCG01AC for Large Systems. Card NTCG02AC is used in Small Systems. These CIS DTI2 cards have increased functionality that allows detection of the CIS toll dial tone.

Feature interactions

CIS 2 Mbit Digital Trunk Interface (CIS DTI2)

CIS DTD feature improves the reliability of outgoing CIS DTI2 toll calls by adding the ability to use the dial tone provided by the CIS CO as criteria to start outpulsing the decadic digits to the toll CIS CO.

Feature packaging

No new package is introduced for this feature. The following packages are required:

- 2 Mbit Digital Trunk Interface (DTI2) package 129
- International Supplementary Features (SUPP) package 131
- Commonwealth of Independent States Trunks (CIST) package 221

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 73 - Add Multifrequency Advanced (MFA) as a CIS firmware type.

2 LD 16 - Specify CIS route information.

LD 73 - Add Multifrequency Advanced (MFA) as a CIS firmware type.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	DTI2	Digital Trunk Interface loops.
FEAT	LPTI	Loop timers and additional definitions.
CDTI2	YES	CDTI2/CSDTI2 card
...	...	
CISFW	MFA	Multifrequency Shuttle protocol handling + ANI Reception + firmware dial tone detection capabilities are supported. MFA capabilities are supported by CIS CDTI2 card (NTCG01AC for Large Systems, or NTCG02AC for Small Systems).

LD 16 - Specify CIS route information.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
TKTP	COT	Central Office Trunk
...	...	...
ICOG	OGT	Outgoing only trunk.
...	...	...
CNTL	(NO) YES	Deny/Allow changes to controls or timers.

- TIMR	ATO 128-(4992)-65408	ANI timeout timer in milliseconds. For CIS outgoing trunk routes this defines the time delay performed after the outpulsing of the toll access code. During this delay further outpulsing is temporarily halted until the special message from the card firmware confirms the satisfaction of the CIS Toll outpulsing criteria. If the timer expires before satisfaction the behavior of the system depends on the COAT prompt. If COAT is set to YES the outpulsing continues on timeout. If COAT is set to NO the call is dropped.
...	...	...
CISR	YES	CIS Route
- DTOC	(NO) YES	Deny/Allow Direct Toll Connection.
- CTOC	(DTDO) AADT ANIO AODT	CIS Toll Outpulsing Criteria. DTD Only. ANI And DTD. ANI Only. ANI Or DTD.
- COAT	(NO) YES	NO = Drop Call when ATO timer expires. YES = Continue Outpulsing when ATO timer expires.

Feature operation

No specific operating procedures are required to use this feature.

CLASS: Calling Number and Name Delivery

Contents

This section contains information on the following topics:

Feature description 1077
Operating parameters 1104
Feature interactions 1105
Feature packaging 1111
Feature implementation 1111
Feature operation 1114

Feature description

The Custom Local Area Signaling Service (CLASS) Calling Number and Name Delivery (CND) feature enables the system to send the calling number and/or calling name to a CLASS set when a call is presented to it, per the Bellcore CLASS CND standard. Once the system delivers the CLASS CND information, it is completely up to the CLASS set to determine how the information is to be displayed. The CLASS set can even choose to ignore certain information by not displaying it.

A CLASS set is, by definition, any non-proprietary analog set with an integrated display and a Frequency Shift Key (FSK) modem receiver, or with a FSK modem receiver built-in display attachment. The CLASS sets are configured on the system as analog (500/2500 type) sets using LD 10, and are supported by the existing 500/2500 type peripheral line cards.

The calling number and/or calling name data is delivered from the system to the CLASS sets using FSK signaling by a CLASS modem (CMOD) unit. The CMOD units are configured using LD 13. They are supported by an Extended CLASS Modem (XCMC) IPE line card, the NT5D60AA.

Up to 255 CMOD units may be configured on a system that is equipped with the CLASS CND feature. Once configured, the CMODs are shared throughout a multi-customer system. When a call is presented to a CLASS set, an available CMOD is automatically allocated.

Upon reaching the CND delivery interval, the appropriate CND information is delivered to the CLASS set. The CND delivery interval is the first silent interval, after ringing has been applied for a new call, that is greater than two seconds.The allocated CMOD unit is released when ringing resumes on the CLASS set after the CND delivery interval, or when the call is disconnected, answered, or redirected. If the call is disconnected, answered, or redirected before the CND delivery interval has been reached, then no CND information is delivered. If the call is disconnected, answered, or redirected while the CND information is being delivered, the CND delivery is immediately terminated. Figures 22 to 25 depict a typical call processing and system resource allocation scenario for a new call being presented on a CLASS set.

Figure 22
System allocation when a CLASS set is idle

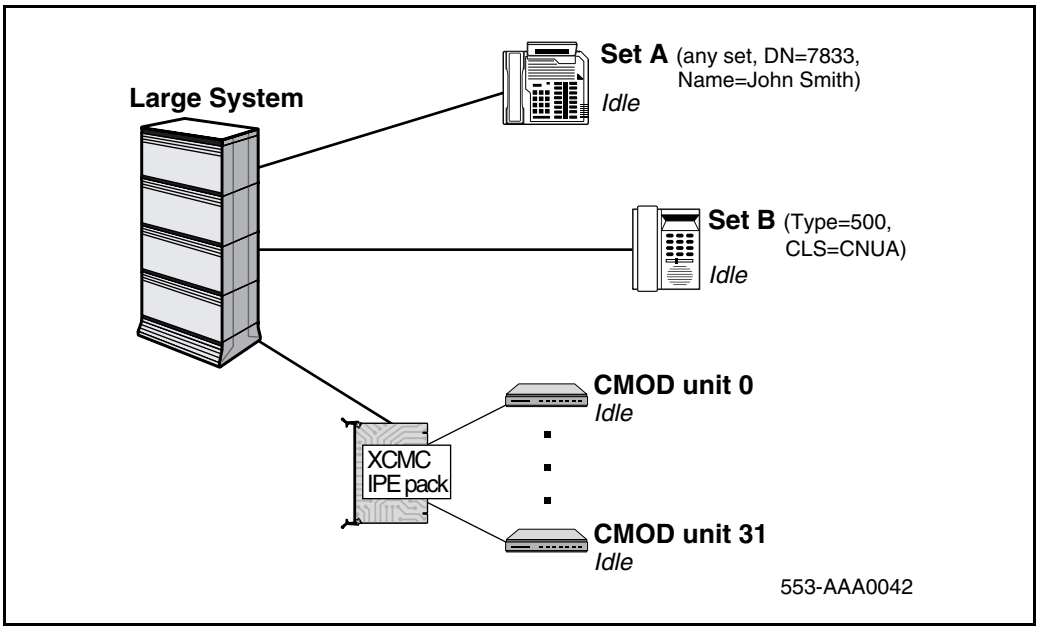


Figure 23
System resource allocation when a new call begins to ring on the CLASS set

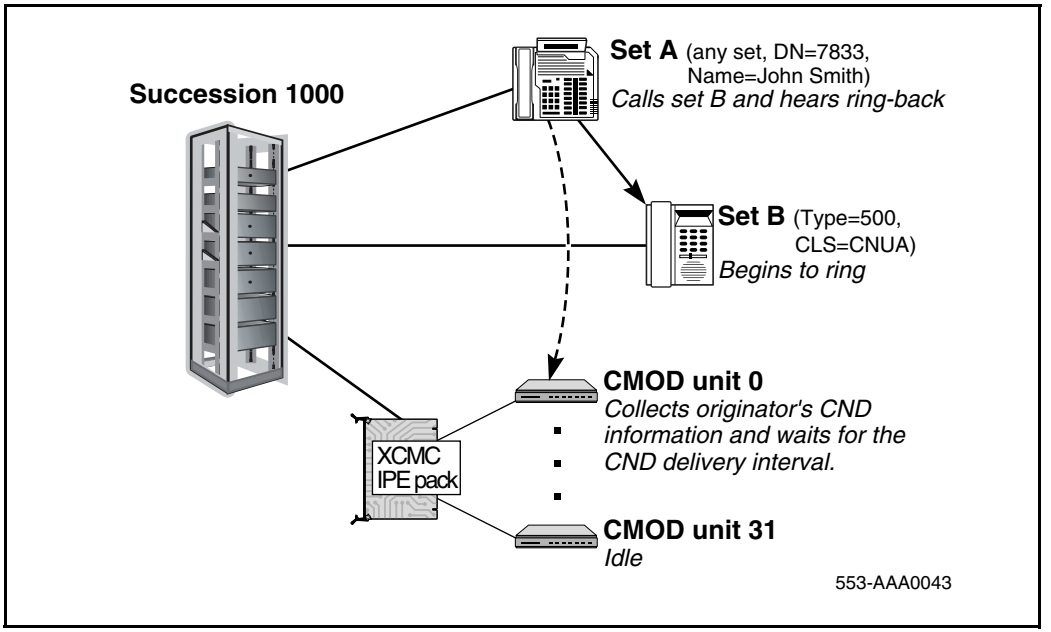


Figure 24
System resource allocation during the CND delivery interval

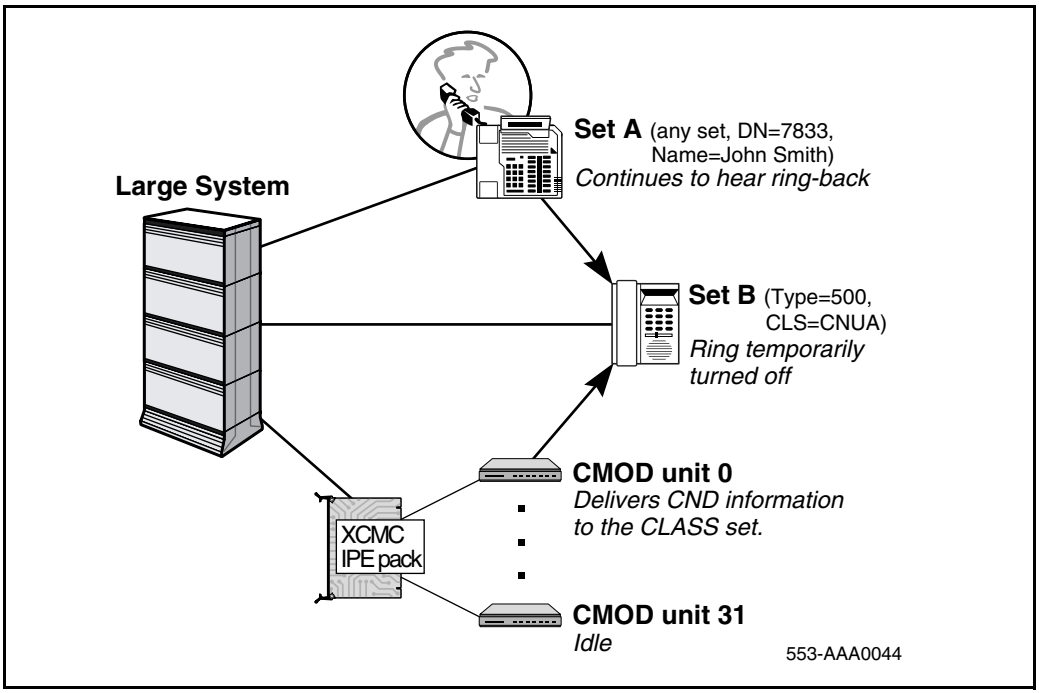
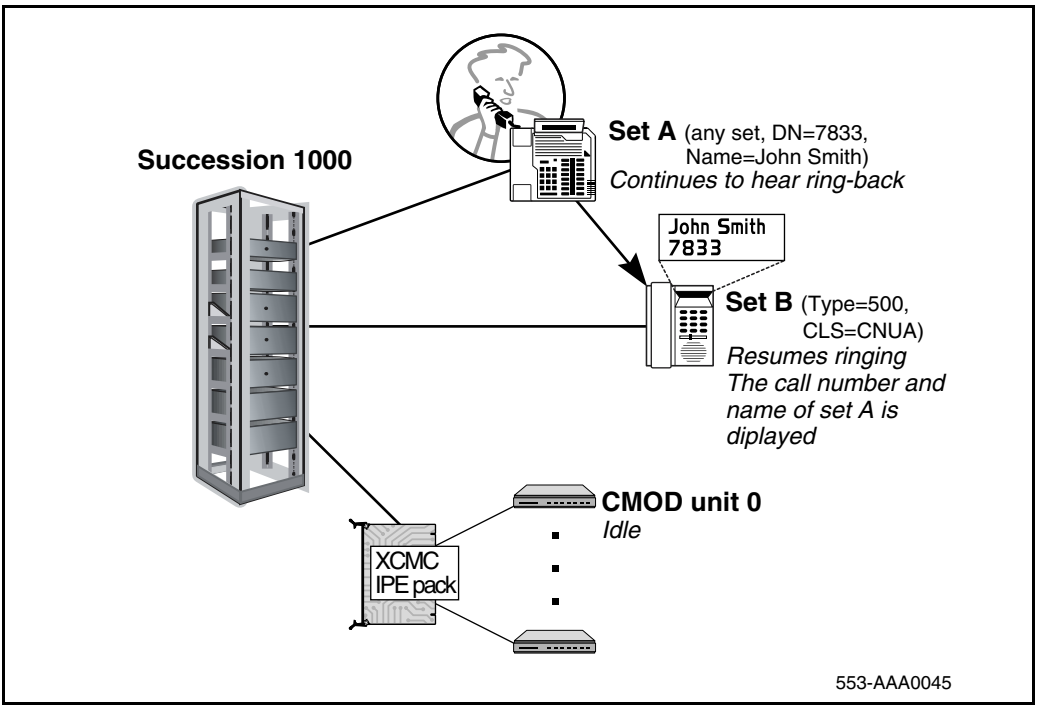


Figure 25
System resource allocation after the CND delivery interval



Configure CND Class of Service on CLASS sets

In LD 10, the system administrator may configure a CLASS set to deliver both Calling Number Delivery, Calling Name Delivery, or both.

To activate only Calling Number Delivery, the station set is to be configured with one of the following class of service:

- CLASS Calling Number Single Data Message Format Allow (CNUS) or
- CLASS Calling Number Multiple Data Message Format Allow (CNUA)

Subsequently, whenever a call is presented to that set, the software will deliver the date and time stamp information (see the Date and time stamp information section which follows below) and the calling number information (see the Calling number information section which follows on page 1084), per the Bellcore CLASS CND delivery standard.

To activate only Calling Name Delivery, the station set is to be configured with the Calling Name Multiple Data Message Format Allow (CNAA) class of service. Subsequently, whenever a call is presented to that set, the system software will deliver the date and time stamp information and the calling name information (see the Calling name information section on page 1085), per the Bellcore CLASS CND delivery standard.

To activate both Calling Number Delivery and Calling Name Delivery, the station set is to be configured with both the CLASS Calling Number Multiple Data Message Format Allow (CNUA) and the CLASS Calling Name Multiple Data Message Format Allow (CNAA) class of service. Subsequently, whenever a call is presented to that set, the system software will deliver the date and time stamp information, the calling number information, and the calling name information, per the Bellcore CLASS CND delivery standard.

Date and time stamp information

The date and time stamp information delivered to a CLASS set is in the format ***mm dd hh mm***, where ***mm*** is a two-digit number for the month, ***dd*** is a two-digit number for the day of the month, ***hh*** is a two-digit number for the military hour of the day, and ***mm*** is a two-digit number for the minute of the hour. It is up to the CLASS set to display the date and time stamp information.

Calling number information

For a non-ISDN trunk-to-CLASS set call, the calling number delivered to the CLASS set will be the incoming In-Band ANI, or if no ANI is passed on, then the calling number unknown indicator is delivered in place of the calling number. The calling number unknown indicator, per Bellcore CLASS CND delivery standard, refers to the ASCII “O” that is sent in place of the calling number.

For an ISDN trunk-to-CLASS set call, the calling number delivered to the CLASS set will be the CLID received from the incoming ISDN trunk (if the CLID is longer than 10 digits, only the first 10 will be delivered). If the incoming CLID is defined as display denied (the originating set has Display Digit Denied, Calling Party Privacy, or Calling Line Identification Restriction active), then the calling number privacy indicator is delivered instead of the calling number, or, if no CLID is passed on by the incoming trunk, then the calling number unknown indicator is delivered in place of the calling number. The calling number privacy indicator, per Bellcore CLASS CND delivery standard, refers to the ASCII “P” that is sent in place of the calling number.

For a station (set)-to-CLASS set call, the calling number delivered to the CLASS set will be one of the following:

- If the originating set has Display Digit Denied active (CLS = DDGD in LD 10 or 11), then the calling number privacy indicator is delivered in place of the calling number.
- If the CLID entry of the originating DN specifies that it is to be identified by its internal DN (CLASS_FMT = DN in LD 15), then the originating DN is delivered.
- If the CLID entry of the originating DN specifies that it is to be identified by its local number (CLASS_FMT = LCL in LD 15), then the local public number associated with the originating DN is delivered, (if the CLID is longer than 10 digits, only the first 10 will be delivered).The local public number is composed by the ISDN CLID Enhancements feature, which forms the calling number by concatenating the customer defined home local number (HLCL in LD 15) and the originating DN.

- If the CLID entry of the originating DN specifies that it is to be identified by its national number (CLASS_FMT = NTN in LD 15), then the national public number associated with the originating DN is delivered (if the number is longer than 10 digits, only the first 10 will be delivered). The national public number is composed by the ISDN CLID Enhancements feature, which forms the calling number by concatenating the customer defined home national number (entered using prompt HNTN and HLCL in LD 15) and the originating DN.
- If the originating set is assigned with an undefined CLID entry, then the originating DN is delivered.

For a station (attendant)-to-CLASS set call, the calling number delivered to the CLASS set will be one of the following:

- If CLID entry 0 specifies that it is to be identified by its internal DN (CLASS_FMT = DN in LD 15), then the customer's attendant DN (the ATDN in LD 15) is delivered.
- If CLID entry 0 specifies that it is to be identified by its local number (CLASS_FMT = LCL in LD 15), then the local public number associated with the customer's Listed Directory Number 0 (the LDN0 in LD 15) is delivered (if the number is longer than 10 digits, only the first 10 will be delivered).
- If CLID entry 0 specifies that it is to be identified by its national number (CLASS_FMT = NTN in LD 15), then the national public number associated with the customer's Listed Directory Number 0 (the LDN0 DN in LD Overlay) is delivered (if the number is longer than 10 digits, only the first 10 will be delivered).
- If CLID entry 0 is not configured, then the customer's attendant DN (the ATDN in LD Overlay) is delivered. The calling number for calls originated by an attendant is composed using CLID entry 0, as per the ISDN CLID Enhancements feature.

Calling name information

For a non-ISDN trunk-to-CLASS set call, the calling name unknown indicator is delivered to the CLASS set in place of the calling name. The calling name unknown indicator, per Bellcore CLASS CND delivery standard, refers to the ASCI "O" that is sent in place of the calling name.

For an ISDN trunk-to-CLASS set call, the calling name delivered to the CLASS set will be the calling name received from the incoming ISDN trunk (if the calling name is longer than 15 characters, only the first 15 will be delivered). If the calling name is defined as presentation denied (the originating set has Display Name Denied, Calling Party Privacy, or Calling Line Identification Restriction active), then the calling name privacy indicator is delivered in place of the calling number, or, if no calling number is passed on by the incoming trunk, then the calling name unknown indicator is delivered in place of the calling name. The calling name privacy indicator, per Bellcore CLASS CND delivery standard, refers to the ASCI "P" that is sent in place of the calling name.

For a station (set)-to-CLASS set call, the calling name delivered to the CLASS set will be one of the following:

- The calling name associated with the originating DN (if the calling name is longer than 15 characters, only the first 15 will be delivered).
- If no name is defined with the originating DN, then the calling name unknown indicator is delivered in place of the calling name.
- If the originating set has Display Name Deny Class of Service (CLS=NAMD in LD 10 or 11), then the calling name privacy indicator is delivered in place of the calling name.

For a station (attendant)-to-CLASS set call, the calling number delivered to the CLASS set will be one of the following:

- The calling name associated with the customer's attendant DN (if the calling name is longer than 15 characters, only the first 15 will be delivered).
- If no name is associated with the customer's attendant DN, then calling name unknown indicator is delivered in place of the calling name.

Network engineering for CLASS sets

In a system with a single group network, the network internal blocking is determined by the concentration ratio of equipped ports on peripheral equipment and the number of interfaced loops or superloops. Depending on traffic engineering, a non-blocking network is achievable.

In a multi-group system, intergroup junctors are required to switch calls between two network groups. Due to the concentration of time slots from a network group to that of inter-group junctors, blocking may occur. This is true for a multi-group system with or without CLASS sets. However, since the CLASS feature depends on a voice path to deliver Calling Name and Number Delivery (CND) to a set, excessive congestion at the inter-group junctor could block the delivery of CND and diminish the usefulness of the feature, as well as impact the grade of service of the existing equipment on the system.

This engineering section examines the inter-group junctor blocking issue and provides recommend engineering rules to alleviate potential network congestion problems.

In general, the engineering effort for CLASS feature can be classified into three categories:

A new site following engineering rules (see page 1094) requires no inter-group junctor traffic check-off.

An existing or new site with relatively low inter-group junctor traffic, will require only one XCMC (Extended CLASS Modem Card) IPE pack that can serve all CLASS sets in a multi-group system.

An existing site with heavy inter-group junctor traffic will require either moving trunks/sets around between network groups when only one XCMC pack is equipped or providing an XCMC pack (or packs) for each group.

System multi-group network

This section is only relevant to a system multi-group network. A single group system does not have inter-group junctors. Therefore, special engineering on the junctor is not applicable. The Succession 1000M Cabinet and Meridian 1 Option 11C Cabinet have network architectures that are different from other systems.

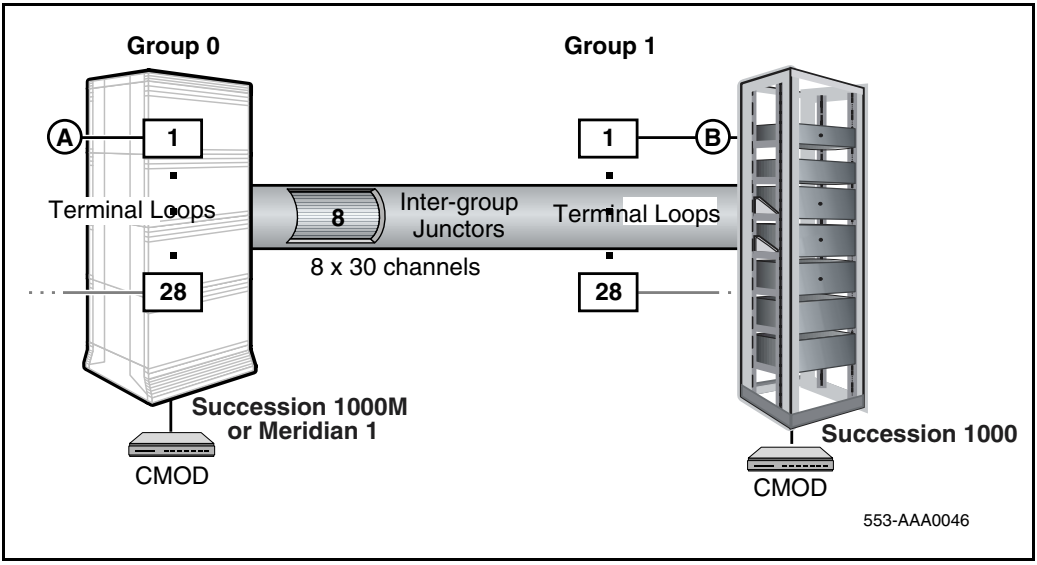
It has a non-blocking network and does not require any network engineering, except to use Table 43 CMOD (Class MODem) capacity table to find the number of modems needed.

In general, inter-group junctor blocking is most severe when there are only two groups, since under typical traffic distribution assumptions, 50% of calls will stay in the originating group and 50% will terminate on the second group through junctors, unless a Community Of Interest (COI) is known and taken into consideration in engineering to minimize inter-group traffic.

Under the assumption of even distribution of traffic, the percentage of traffic to an inter-group junctor will drop to 33.3% of the total group traffic for a three-group system. Similarly, the junctor traffic will be 25% of group traffic for a four-group system and 20% for a five-group system.

A simplified system with two-group network and CLASS sets is shown by Figure 26.

Figure 26
A system with a two-group network



Note that traffic to a CLASS set can be originated from a non-CLASS set, another CLASS set or an incoming trunk. Since trunks have more traffic impact on junctor blocking, they are used to illustrate the feature operation, however, both sets and trunks can be traffic sources to CLASS sets.

The maximum size system comprises of 5 network groups. Each group has 32 loops of which 28 can be terminal loops, the rest are service loops (TDS loops for tones, conference and music).

From Group 0 to Group 1, as shown in Figure 26, there are 8 one-way junctors. Similarly, there are another 8 one way junctors from Group 1 to Group 0. For practical purposes, they can be treated as 8 two-way junctors. A two-way path is equal to a voice channel. A junctor has 30 voice traffic channels as on a loop. Each two-way channel represents a conversation path. A channel is also used to deliver the CND from the CMOD to a CLASS set.

CLASS feature operation

A call originated from Set A (or trunk A) seeks to terminate on a CLASS set B. When B starts to ring, A will hear ringback. A unit in CMOD (CLASS Modem) is assigned to collect originator's CND information and waits for the CND delivery interval. After the first ring at B, a silence period (deliver interval) ensues, the CMOD unit begins to deliver CND information to the CLASS set.

The CND information of a traffic source (A) is a system information, which is obtained by the system when a call is originated. During the two-second ringing period of the CLASS set B, A's CND is delivered to CMOD through SSD messages (using a signaling channel only). When the CND information is sent from CMOD to CLASS set B, it is delivered through a voice path during the four-second silence cycle of set B. The CMOD unit is held for a duration of six seconds.

If the XCMC (Extended CLASS Modem Card) IPE pack, which provides up to 32 CMOD units, is located in the IPE of Group 0, the CMOD unit in the pack will receive CND data through the SSD messages and use one of the voice channels of the inter-group junctor to deliver it to CLASS set B in Group 1.

If the XCMC IPE pack is located in Group 1, the system will deliver SSD messages containing CND information to CMOD and then send it to Set B during the delivery interval through a voice path, which is an intra-group channel not involving an inter-group junctor.

When CMOD units and CLASS sets are co-located in the same network group, there are no voice paths on the inter-group junctor required to deliver CND information; when they are equipped on different groups, inter-group junctors must carry CND traffic. The resource allocation algorithm will search for a CMOD unit located in the same group as the terminating CLASS set first before it attempts to use a CMOD unit from a different group.

Inter-group junctor capacity

The time slot allocation algorithm for inter-group junctors is somewhat different from regular network loops, however, it is a close approximation to use the network loop capacity for junctor capacity, since they each have 30 traffic channels.

In order for the connection between a CMOD unit and the remote set (on a different group) to meet the Grade Of Service (GOS) of the system for internal call setup, traffic on the loop and inter-group junctor should meet normal GOS requirements, that is 660 CCS per loop or junctor.

If we let an inter-group junctor be loaded to 660 CCS like a loop, the total allowed traffic at eight junctors is 5280 CCS (=660x8). At 6 CCS per CLASS set, the allowed number of sets generating inter-group traffic is 880 (=5280/6). If half of set traffic is intra-group, and the other half inter-group, the allowed number of CLASS sets in Group 1 is 1,760 (=880/0.5). On per loop basis, each loop can be equipped with 62 (=1760/28) CLASS sets.

Similarly, a 3-group network is likely to have 1/3 of traffic remaining in the group, 1/3 of traffic going to the next group, and another 1/3 of calls going to the third group. By using this inverse proportion approach to breakdown traffic flow at junctors, the capacity of a network group in terms of CLASS sets is summarized in Table 42. The entry in the Table 42 is the threshold for inter-group junctor traffic check-off.

The table entry indicates that in a two group system, the second group is allowed to have 1760 CLASS sets or a combination of CLASS and equivalent sets without a need to move sets or trunks around to reduce inter-group junctor traffic. Due to higher traffic, an agent set or a trunk is counted to be equal to four regular sets. The conversion needs to be done before using Table 42.

If the number of equipped sets (CLASS or otherwise) or equivalent sets (sets converted from trunks and agent sets) is less than the threshold, the junctor traffic is expected to be low. There is no special engineering necessary for CLASS sets (other than providing required modems).

Note that the junctor traffic issue can be ignored if each group is fully equipped with sufficient CMOD units to handle CLASS sets within the group. It is the attempt to use one XCMC pack to serve multiple groups that requires special attention to inter-group traffic. As long as CLASS service is not impacted by a traffic in-balance already existed in the system, it is not the objective of this engineering guideline to solve that problem.

Once the threshold is exceeded, re-configuration of the system is necessary to reduce junctor congestion. The detailed engineering rules are given later in this document.

Table 42
Maximum CLASS sets per group (based on inter-group junctor capacity limitation)

No. of Groups	Sets/Group
2	1760
3	2933
4	3520
5	4080

Note: Convert a trunk or an agent set to four equivalent sets before applying Table 42.

A single group system can have as many CLASS sets as each loop allows. The engineering of the system is not different from that of non-CLASS sets, since there is no inter-group junctor involved. The only engineering required is to find the required number of CMOD units from Table 43 to serve a given number of CLASS sets.

Note that the capacity per group for multigroup systems assumes no trunking in that particular group (or trunks have been converted to equivalent sets). Therefore, the total system capacity, by taking into account trunks, agent sets and service circuits, will not be as large as a straight multiple of number of groups by the number of sets per group.

The number of allowed CLASS sets per group in Table 42 is strictly a function of inter-group traffic (except for five-group systems). When a system becomes five-group, the junctor capacity is no longer a bottleneck under the assumption of even traffic distribution. The system capacity will become unrealistic if it is purely based on the inter-group junctor capacity, therefore, other system resources, particularly the system CPU, need to be checked. The number 4080 is based on loop traffic (28 loops/7 superloops), not junctor capacity.

If a group comprises of both regular sets and CLASS sets, the total number of sets in the group should not exceed the quoted number in the table. If trunks and agent sets are included in the group, convert them to "equivalent sets" before using Table 42. More details are described in the engineering guide.

Table 43 shows the CMOD capacity. It provides the number of CMOD units required to serve a given number of CLASS sets with the desired grade of service. The required number of CMOD units should have a capacity range whose upper limit is greater than the number of CLASS sets equipped in a given configuration.

The procedure to use Table 43 is further illustrated in engineering examples starting on page 1096.

Table 43
CMOD unit capacity

CLASS Set	1-2	3-7	8-27	28-59	60-100	101-150	151-200	207-267
CMOD Unit	1	2	3	4	5	6	7	8
CLASS Set	268-332	333-401	402-473	474-548	549-625	626-704	705-785	786-868
CMOD Unit	9	10	11	12	13	14	15	16

CLASS Set	869-953	954-1039	1040-1126	1127-1214	1215-1298	1299-1388	1389-1480	1481-1572
CMOD Unit	17	18	19	20	21	22	23	24
CLASS Set	1573-1665	1666-1759	1760-1854	1855-1949	1950-2046	2047-2142	2143-2240	2241-2338
CMOD Unit	25	26	27	28	29	30	31	32
CLASS Set	2339-2436	2437-2535	2536-2635	2637-2735	2736-2835	2836-2936	2937-3037	3038-3139
CMOD Unit	33	34	35	36	37	38	39	40
CLASS Set	3140-3241	3242-3344	3345-3447	3448-3550	3551-3653	3654-3757	3768-3861	3862-3966
CMOD Unit	41	42	43	44	45	46	47	48
CLASS Set	3967-4070	4071-4175	4176-4281	4282-4386	4387-4492	4493-4598	4599-4704	4705-4811
CMOD Unit	49	50	51	52	53	54	55	56
CLASS Set	4812-4918	4919-5025	5026-5132	5133-5239	5240-5347	5348-5455	5456-5563	5564-5671
CMOD Unit	57	58	59	60	61	62	63	64

General engineering guidelines for non-Call Center applications

In a non-call center application, there is no significant number of agent sets. Therefore, no agent set to regular set conversion is needed. The only type of port requiring special treatment is trunk.

Configurations following engineering rules (no re-configuration required)

The following engineering rules should be followed to avoid the need to re-configure a switch to accommodate the CLASS feature.

1 Provide the number of CMOD units serving all CLASS sets in the system based on the capacity table (Table 43).

2 Equip all CLASS sets in one network group.

3 Equip the XCMC IPE pack on the network group with CLASS sets.

If the system is a single group system, or if above rules are fully met, no further engineering is necessary. However, in case of an existing multi-group site upgrading to provide CLASS feature, we may need to re-configure the system in order to satisfy rule (2).

When the above rules are not fully met, continue the system engineering by following the procedure in the next subsection.

Re-configuration may be required (when engineering rules are not fully followed)

When above rule (2) can not be satisfied in a new site or an existing one, the following guidelines are designed to minimize network blocking, and to determine whether a re-configuration (to move trunks and sets around) or to provide an XCMC pack per group is necessary.

1 To use Table 43 to estimate CMOD unit requirements, consider only CLASS sets (no trunks or non-CLASS sets).

2 If CLASS sets are equipped in more than one group, locate the XCMC IPE pack in the group with most CLASS sets.

3 Use Table 42 to decide whether re-configuration is required. For a network group with trunks, regular sets and CLASS sets, convert trunks to sets by using the formula: 1 trunk = 4 sets (called equivalent sets), and then add up the total.

Check threshold in Table 42, if the number of equivalent sets is less than 1760 (for example, for a two-group system), there is no need to re-configure the system.

If the number is greater than 1760, we need to move some of the CMOD units to a second XCMC IPE pack on another group (when CLASS sets are scattered in two groups), or move some sets or trunks from one group to another group to satisfy the threshold.

The following examples will show some of the engineering details of dealing with various alternatives.

To simplify discussion, the network group 0 has minor number of CLASS sets. The majority of CLASS sets are in group 1 (refer to Figure 26).

Engineering examples for non-Call Center applications

One XCMC pack serving a single group system

No special engineering rule is needed for a single or half group system (Succession 1000M Single Group, Meridian 1 Option 61C CP PII, and Succession 1000M Half Group). Look up Table 43 to find the required number of CMOD units to serve the given CLASS sets. For example, to serve a Succession 1000M Single Group with 400 CLASS sets, use Table 43 to find the number of CMOD units serving a range including 400 sets. The result is 10 units which can serve 333 to 401 CLASS sets.

One XCMC pack serving a 2-group system

1 **Example 1**: No re-configuration

A 2-group system serving an office is expected to convert 400 analog sets to CLASS sets. Currently, 100 of them are located in group 0, where all incoming trunks are located, and the remaining 300 sets are in group 1. Assume that group 1 is also equipped with 800 non-CLASS sets. How many CMOD units are needed to serve this application and does the customer need to re-configure the switch (move sets and trunks between group 0 and group 1) to do the upgrade?

Solution:

The table lookup indicates that 400 CLASS sets need 10 CMOD units. Since one pack provides 32 units, one XCMC pack is sufficient for this customer.

Group 1 is equipped with 300 CLASS sets which is greater than the 100 sets in group 0, the pack should be installed in group 1.

The total equipped ports in group 1 is 1100 (=800+300). For a 2-group system, the second group is allowed to have 1760 sets (from Table 42) without junctor traffic concerns, therefore, there is no need for the customer to re-configure the switch.

2 **Example 2**: Re-configuration

A similar application as in the last example, except that there are 1600 non-CLASS sets and 100 trunks in group 1.

Solution:

The same number of CMOD units (10), since the number of CLASS sets in the system is the same.

The number of total equivalent sets in group 1 is 2300 (=1600 +300 +100x4) which is greater than the 1760 threshold in the Table 42 for a 2-group system.

The customer will have a number of alternatives to resolve the junctor blocking issue, depending on the situation:

(1) move the 100 CLASS sets from group 0 to group 1, so all CLASS sets are served by the XCMC pack in group 1, or

(2) move the 300 CLASS sets and the XCMC pack to group 0, or

(3) move 540 non-CLASS sets (=2300-1760) from group 1 to group 0, or

(4) move 100 CLASS sets from group 1 to group 0 and split the 10 CMOD units to 5 for group 0 and 5 for group 1. However, this will require another XCMC pack to be equipped in group 0. The cost of this approach is not trivial. It can be justified only when growth plan indicates a need for a second pack in the near future anyway.

The final decision depends on the specific situation of a site.

General engineering guidelines for Call Center applications

Configurations following engineering rules (no re-configuration required)

The following engineering rules should be followed to avoid the need to re-configure a switch to accommodate the CLASS feature for call center environment.

1 Convert an agent set to regular set by using 1 agent CLASS set = 4 sets (called equivalent sets)

2 Sum up the total number of regular CLASS sets and equivalent CLASS sets and find the number of CMOD units required based on the capacity table (Table 43).

3 Equip CLASS agent sets in the group where trunks carrying incoming traffic to agent sets are located.

4 Equip non-agent CLASS sets in the same group as the agent CLASS sets.

5 Equip the XCMC IPE pack on the network group with CLASS sets.

If the system is a single group system, or if above rules are fully met, no further engineering is necessary.

In case of an existing multi-group site upgrading to provide CLASS feature, re-configuring the system may be necessary to satisfy rules (3) and (4).

When above rules are not fully met, continue the system engineering by following the procedure in the next subsection.

Configurations do not fully meet engineering rules (re-configuration may be required)

When above rules can not be satisfied in a new site or an existing one, the following guidelines are designed to (1) minimize network blocking, (2) determine whether a re-configuration (to move trunks and sets around) is necessary, or (3) whether separate XCMC packs are necessary to serve the multi-group system.

1 Convert an agent set to regular set by using 1 agent CLASS set = 4 sets.

2 Sum up the total number of equivalent CLASS sets and find the number of CMOD units required based on the capacity table (Table 43).

3. Equip the XCMC IPE pack on the network group with most CLASS sets (or equivalent sets).
4. Limit the number of agent CLASS sets to be 200 or less per group.
5. Limit the number of regular CLASS sets in a group without XCMC pack to be 100 or less.
6. Convert trunks (1 trunk = 4 sets), and agent set (1 agent set = 4 sets) to equivalent sets before using Table 42 to find junctor traffic threshold.
7. If the threshold in Table 42 is greater than the total number of equivalent sets, traffic in the system is balanced, there is no need for further network engineering.
8. If the number of equivalent set is greater than the threshold, one or all of the following engineering rules should be followed to reduce junctor traffic:
   a. Move sets (CLASS or non-CLASS) or trunks to another group to satisfy the above engineering rules.
   b. Equip XCMC pack in more than one group to serve local CLASS set traffic.
9. When a trunk route is known to serve only agent sets, and these trunks and agent sets are in the same group, exclude them from the set count in Table 42 threshold (for example, do not include trunks and agent sets with known COI to use Table 42; they do not generate traffic to junctors).

The following examples will show some of the engineering details of dealing with various alternatives.

To simplify discussion, the network group with most trunks is called group 0, consequently, a majority of CLASS sets, if not all, are in group 1. If most agent CLASS sets and XCMC pack are in group 0, there will ne no need for further engineering.

Engineering examples for Call Center applications

One XCMC pack serving a single group system

No special engineering rule is needed for a single group system. Look up Table 43 to find the required number of CMOD units to serve the given CLASS sets. For example, to serve a Succession 1000M Single Group or Meridian 1 Option 61C CP PII with 300 agent CLASS sets, use Table 43 to find the CMOD units that can serve 1200 equivalent sets (=300x4). The result is 20 units.

One XCMC pack serving a 2-group system

1 **Example 1**: No re-configuration required

A two-group system serving a call center is expected to upgrade 300 analog sets (100 administrative sets and 200 agent sets) to CLASS sets. The 100 administrative sets are located in group 0, where are also located all incoming trunks. The 200 agent sets are in group 1, which will continue to be used as agent sets after upgrading. Assume that group 1 is also equipped with 500 non-CLASS sets. How many CMOD units are needed to serve this application and does the customer need to re-configure the switch (move sets and trunks between group 0 and group 1) to do the CLASS feature upgrade?

Solution:

The table lookup indicates that 900 equivalent CLASS sets (=100+200x4) need 17 CMOD units. Since one pack provides 32 units, one XCMC pack is sufficient for this customer.

Group 1 is equipped with 200 agent CLASS sets or 800 equivalent sets which is greater than the 100 sets in group 0, the pack should be installed in group 1.

The total equipped ports in group 1 is 1300 (=200x4+500). For a 2-group system, the second group is allowed to have 1760 sets (from Table 42) without junctor traffic concerns, therefore, there is no need for the customer to re-configure the switch.

In addition, both “100 CLASS sets in a group without CMOD units (group 0)”, and” 200 agent CLASS sets in a group separate from incoming trunks (group 1)” statements are within engineering rules, therefore, no re-configuration is necessary.

2 **Example 2**: Re-configuration required

A similar application as in the last example, but there are 1600 non-CLASS sets in group 1.

Solution:

The same number of CMOD units (17) is required, since the number of equivalent CLASS sets (900) in the system is the same.

Equip the 17 CMOD units in group 1, since the XCMC pack should be equipped in the group with most CLASS (equivalent) sets.

The number of total equivalent sets in group 1 is 2400 (=1600+200x4) which is greater than the 1760 threshold in Table 42 for a two-group system.

The customer will have a number of alternatives to resolve the junctor blocking issue, depending on the situation:

Move the 100 CLASS sets from group 0 to group 1, or

Equip the 200 CLASS agent sets and the XCMC pack in group 0, or

Move 640 non-CLASS sets (=2400-1760) from group 1 to group 0 to avoid threshold violation, or

Move 100 CLASS sets from group 1 to group 0 and split the 17 CMOD units to 10 for group 0 and 7 for group 1; however, this will require another XCMC pack to be equipped in group 0, or

Move 160 trunks with COI to agent sets from group 0 to group 1, so that the total equivalent sets in group 1 will become 1760 (=1600+(200-160)x4), since this way the 160 trunks and an equal number of agent sets will not generate traffic to junctors.

The final decision depends on the specific situation of a site.

3 **Example 3**: Mixed sets, trunks in both groups and re-configuration required

A two-group system serving a call center is expected to equip 200 administrative CLASS sets in group 0 and 400 CLASS agent sets in group 1. 500 trunks carrying incoming traffic to agents are located in group 0, 60 trunks serving local CO non-ACD traffic are equipped in group 1. Assume that group 1 is also equipped with 300 non-CLASS sets. Can this configuration meet engineering rules? How many CMOD units are needed?

Solution:

The equivalent CLASS sets in system = 200 + 400x4 = 1800. From Table 43, 27 CMOD units are needed. It requires the XCMC pack to be equipped in group 1.

When we equip the XCMC pack in group 1, there are three violation of rules: (1) the number of CLASS sets in the group without CMOD units (group 0) is greater than 100, (2) the number of agent sets in a group without incoming trunks (group 1) is 400 which exceeds the 200 per group limit, and (3) the violation of threshold in Table 42 for group 1 (=400x4+60x4+300=2140>1760). Several alternatives are available to make this configuration meeting engineering rules:

Move 100 CLASS sets and 400 incoming trunks from group 0 to group 1; all above 3 violations are removed by this re-configuration: (1) CLASS sets in group 0 is 100, (2) 400 CLASS agent sets and 400 incoming trunks with COI are in the same group (group 1), (3) the number of equivalent sets in group 1 for threshold check-off is reduced to 640 (=100+60x4+300=640) which is certainly within the limit (1760).

However, it is impractical to put almost all trunks and agent sets in one group (group 1). With so many rule violations, the most realistic approach is to move 200 CLASS agent sets to group 0 and equip approximately 15 CMOD units in group 0 and 12 units in group 1.

One XCMC pack serving a 3-5 group system

Chances of groups larger than three requiring special engineering are slim, since the threshold (Table 42) limiting the number of sets per group is much higher.

If the rule of co-locating CLASS sets and CMOD units in the same group is not fully met, as long as basic rule like putting XCMC pack in the group with most CLASS sets is followed, perhaps, no re-configuration between any two groups is necessary.

However, if in doubt, isolate any two groups at one time, and go through the two-group engineering procedure to re-configure the system two groups at a time. Ignore the rest of system during the engineering process, except for calculating the total number of CMOD units, which should cover the need of all CLASS sets in the system. However, during a two-group engineering, only the number of CMOD units attributable to the two-group at hand should be used in calculations.

Also remember to use 2933 (equivalent) sets per group for threshold check-off for the three-group system, and their respective number for four- and five-group systems (Table 42).

The complete check-off of set threshold between any two groups in a multi-group system can be represented by the following combinations (a number denotes the group number: for example, 1-2 represents group one-group 2):

three-group: 1-2, 1-3, 2-3.

four-group: 1-2, 1-3, 1-4, 2-3, 2-4, 3-4.

five-group: 1-2, 1-3, 1-4, 1-5, 2-3, 2-4, 2-5, 3-4, 3-5, 4-5.

It should be noted that although CMOD units are equipped according to the traffic requirement of CLASS sets in a network group for the inter-group junctor traffic consideration, they are a system resource shared by the whole system.

Operating parameters

Dialed Number Identification System (DNIS) is not supported by the CLASS feature.

This feature has been developed for the North American residential market, as well as for the small business, hospitality, schools and universities, nursing homes and hospitals, and mobile home markets.

This feature applies to a standalone environment, and to an ISDN networking environment supporting CLID and a non-ISDN networking environment with or without In-Band Automatic Number Identification (ANI).

If the CLASS Calling Number Delivery and CLASS Calling Name Delivery data to a CLASS set is lost, there is no indication, such as an error message, provided. The call is presented in the normal manner, as if the CLASS CND function has not been activated.

A maximum of 10 digits can be delivered as the calling number, and a maximum of 15 characters can be delivered as the calling name. These are Bellcore limitations.

The 16 port Analog Message Waiting line card (QPC789A) is not supported for CLASS: Calling Number and Name Delivery.

Up to 255 CLASS modems can be configured on a system, and up to 32 CLASS modem units can be configured on any Extended CLASS Modem card.

No CLASS CND traffic measurement is supported.

The CND delivery interval is the first silent interval, after ringing has been applied for a new call, that is greater than two seconds. If the ringing applied to the CLASS set uses a cadence with a ring-off cycle that is more than four seconds in duration, then the CLASS set might view the call as having been disconnected and might clear the CND display as a result. This is a CLASS set firmware limitation.

Due to firmware limitations on some types of CLASS sets, after a call has been disconnected, the CND information associated with the disconnected call may still appear on the CLASS set's display for several seconds. During this time, the CLASS set cannot display new CND information. Though the system delivers the CND information pertaining to a new call, there is no guarantee that the CLASS set will display the delivered information.

Feature interactions

Attendant Call Extension

When an attendant extends a call to a CLASS set, the Calling Name and Number of the attendant is delivered to the CLASS set, and not that of the extended station.

Automatic Wake Up

When an Automatic Wake Up call is presented to a CLASS set with Calling Name Delivery activated, then the calling name unknown indicator is delivered in place of the calling number.

Blind Transfer

When a Meridian 1 proprietary set completes a Blind Transfer to a CLASS set, the Calling Name and Number of the transferring set is delivered to the CLASS set, and not that of the set being transferred. When an analog (500/2500 type) set completes a Blind Transfer to a CLASS set, the transfer is presented as a new call to the CLASS set. Therefore, the Calling Name and Number of the transferred set is delivered to the CLASS set.

Call Forward All Calls

When a call is redirected to a CLASS set using Call Forward All Calls, the Calling Name and Number of originating set, and not that of the forwarding set, is delivered to the CLASS set.

Call Forward, Internal

When a call is redirected to a CLASS set using Internal Call Forward, the Calling Name and Number of originating set, and not that of the forwarding set, is delivered to the CLASS set.

Call Transfer

When a Call Transfer has been completed, the Calling Name and Number of the transferred party will not be delivered.

Calling Line Identification Restriction for ISDN BRI sets

When an ISDN BRI set, with Calling Line Identification Restriction active, makes a nodal call to a CLASS set with Calling Name Delivery active, the calling name privacy indicator will be delivered as the calling name.

Conference/No Hold Conference

When a set initiates a conference call to a CLASS set, the Calling Name and Number of the initiating set is delivered to the CLASS set.

Dial Intercom

The CLASS Calling Name and Number class of service cannot be configured on Dial Intercom Group sets.

Direct Inward System Access (DISA)

If a call is made to a CLASS set using DISA dialing, then the Calling Name and Number delivered to the CLASS set is that of the incoming trunk and not the DISA DN. This is consistent with the normal display function for DISA calls.

Display of Calling Party Denied

When a set with Display of Calling Party Denied active makes a nodal call to a CLASS set with Calling Number Delivery active, then the calling name privacy indicator is delivered in place of the calling name.

Distinctive Ringing

The normal delivery cycle for Calling Name and Number Delivery applies to calls presented to a CLASS set with distinctive ringing, that is, the Calling Name and Number Delivery information is delivered during the first silent period that is longer than two seconds.

Flexible Numbering Plan (FNP) Enhancement

For a customer equipped with the FNP package 160, the calling number delivered to the CLASS set can be of any length, up to 10 digits.

For a customer not equipped with FNP, the following apply:

- For a station (set or attendant) making a call to a CLASS set, if the local public number is to be delivered to the CLASS set as the calling number of the calling station, then the system software will pad or truncate the calling number to be exactly seven digits long.
- For a station (set or attendant) making a call to a CLASS set, if the national number is to be delivered to the CLASS set as the calling number of the calling station, then the system software will pad or truncate the calling number to be exactly 10 digits long.
- For an incoming trunk call to a CLASS set, the incoming Calling Number Identification/Automatic Number Identification that is passed to the CLASS set be of any length, up to 10 digits.

Group Call

When a set makes a Group Call to a CLASS set with Calling Number Delivery active, the calling number unknown indicator will be delivered in place of the calling number.

When a set makes a Group Call to a CLASS set with Calling Name Delivery active, the calling name unknown indicator will be delivered in place of the calling name.

Hotline

When a Hotline call is made to a CLASS, then the Calling Name and Number is delivered in the same manner as when a call is presented to the CLASS set using normal dialing.

Hunt

When a call is redirected to a CLASS set using Hunt, the Calling Name and Number of originating set, and not that of the redirecting set, is delivered to the CLASS set.

Incremental Software Management (ISM)

While no new ISM limit is introduced by the CLASS feature, each CLASS modem unit TN is counted against the system TN limit and is reflected in the overlay banner.

Feature Group D

When an incoming Feature Group D trunk call is presented to a CLASS set with Calling Number Delivery active, the calling number delivered to the CLASS set will be one of the following:

- If the Feature Group D trunk route is configured so as to not show the incoming ANI (SHAN = NO in LD 19), the calling number privacy indicator is delivered in place of the calling number.
- If no ANI is passed on, then the calling number unknown indicator is delivered in place of the calling number.
- Otherwise, the incoming Feature Group D trunk ANI is directly delivered.

In-Band ANI

When an incoming In-Band ANI trunk call is presented to a CLASS set with Calling Number Delivery active, the calling number delivered to the CLASS set will be one of the following:

- If no ANI is passed on, then the calling number unknown indicator is delivered in place of the calling number.
- Otherwise, the incoming In-Band ANI is directly delivered.

Integrated Services Digital Network

When an incoming Integrated Services Digital Network (ISDN) trunk call is presented to a CLASS set with Calling Number Delivery active, the calling number delivered to the CLASS set will be one of the following:

- If the incoming CLID is defined as display denied (the originating set has Display Digit Denied, Calling Party Privacy, or Calling Line Identification Restriction active), then the calling number privacy indicator is delivered in place of the calling number, or, if no CLID is passed in from the incoming trunk, then the calling number unknown indicator is delivered in place of the calling number.

- If no CLID is passed on by the incoming trunk, then the calling number unknown indicator is delivered in place of the calling number.
- Otherwise, the CLID received from the incoming ISDN trunk is directly delivered. If the CLID is longer than 10 digits, only the first 10 will be delivered.

When an incoming ISDN trunk call is presented to a CLASS set with Calling Name Delivery active, the calling name delivered to the CLASS set will be one of the following:

- If the calling name is defined as presentation denied (the originating set has a Display Name Denied, Calling Party Privacy, or Calling Line Identification Restriction active), then the calling name privacy indicator is delivered instead of the calling name.
- If no calling name is passed on by the incoming trunk, then the calling name unknown indicator is delivered in place of the calling name.
- Otherwise, the calling name received from the incoming ISDN trunk is directly delivered. If the calling name is longer than 15 characters, only the first 15 will be delivered.

Meridian 911

When an incoming M911 trunk call is presented to a CLASS set with Calling Number Delivery active, the calling number delivered to the CLASS set will be one of the following:

- If no ANI is passed on, then the calling number unknown indicator is delivered in place of the calling number.
- Otherwise, the incoming ANI is directly delivered.

Private Line Service

When a Private Line Service call is made to a CLASS, then the Calling Name and Number is delivered in the same manner as when a call is presented to the CLASS set using normal dialing.

VIP Automatic Wakeup

When an attendant makes a VIP Automatic Wakeup call to a CLASS, then the Calling Name and Number is delivered in the same manner as when the attendant makes a call to the CLASS set using normal dialing.

Virtual Network Services (VNS)

When an incoming VNS trunk call is presented to a CLASS set with Calling Number Delivery active, the calling number delivered to the CLASS set will be one of the following:

- If the incoming CLID is defined as display denied (the originating set has Display Digit Denied, Calling Party Privacy, or Calling Line Identification Restriction active), then the calling number privacy indicator is delivered in place of the calling number.
- If no CLID is passed on by the incoming VNS trunk, then the calling number unknown indicator is delivered in place of the calling number.
- Otherwise, the CLID received from the incoming VNS trunk is directly delivered. If the CLID is longer than 10 digits, only the first 10 will be delivered.

When an incoming VNS trunk call is presented to a CLASS set with Calling Name Delivery active, the calling name delivered to the CLASS set will be one of the following:

- If the calling name is defined as presentation denied (the originating set has a Display Name Denied, Calling Party Privacy, or Calling Line Identification Restriction active), then the calling name privacy indicator is delivered in place of the calling name.
- If no calling name is passed on by the incoming VNS trunk, then the calling name unknown indicator is delivered in place of the calling name.
- Otherwise, the calling name received from the incoming VNS trunk is directly delivered. If the calling name is longer than 15 characters, only the first 15 will be delivered.

Feature packaging

The following packages are required for the CLASS Calling Name and Number Delivery feature:

- Calling Party Name Display (CPND) package 95
- CLASS Calling Number Delivery (CNUMB) package 332
- CLASS Calling Name Delivery (CNAME) package 333 and

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 - Configure the CLASS CLID parameters in the Customer Data Block.

2 LD 10 - Configure the CLASS Calling Name and Calling Number Delivery Class of Service for Analog (500/2500 type) sets.

3 LD 13 - Configure the CLASS modem unit (up to 255 CLASS modem units may be configured per system).

LD 15 - Configure the CLASS CLID parameters in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change or delete existing data.
TYPE:	NET	Networking data.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

CLID		CLID option.
	(NO)	NO = (the default) do not configure a CLID table. In this case, the remaining prompts are not generated, and no CLID is sent for the customer.
	YES	YES = configure a CLID table for the customer.
- SIZE	0-(256)-4000	The maximum number of CLID entries that are required.
...		
- ENTRY	aaaa Xaaaa Xaaaa Xbbbb <CR>	aaaa = CLID entry to be configured. Xaaaa = CLID entry to be deleted. Xaaaa Xbbbb = CLID entries to be deleted. aaaa and bbbb must be a value between 0 and (SIZE-1). The action for the entry will be saved to system memory after the CLID entry has been completely configured. If an existing CLID entry is changed, the message "ENTRY aaaa SAVED" is displayed. If a CLID entry or CLID entries is/are deleted, the message "ENTRY aaaa DELETED" or "ENTRIES aaaa-bbbb DELETED" is displayed.
...		
- LSC	0-9999999 X	Local steering code, 1-7 digits. X = delete digits.
- - CLASS_FMT	(DN) LCL	Send internal DN to a CLASS set as the calling number. Send local number to a CLASS set as the calling number.
	NTN	Send National Number to a CLASS set as the calling number.
ENTRY aaaa SAVED ENTRY aaaa DELETED ENTRIES aaaa-bbbb DELETED		Displayed message. Refer to the ENTRY prompt description.
...		

Note 1: You may print the CLASS_FMT information included in the CLID using LD 21.

Note 2: You may print or count the station sets class of service using LD 81. This information would typically be used for billing purposes, to bill set users for calls according to the assigned class of service.

LD 10 - Configure the CLASS Calling Name and Calling Number Delivery Class of Service for Analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
DN	x..x yyyy	Directory Number for this set (x..x) and CLID entry associated with this set (ENTRY configured in LD 15).
	(CNUD) CNUA CNUS (CNAD) CNAA	CLASS Calling Number Delivery Denied. CLASS Calling Number Multiple Data Format Allowed. CLASS Calling Number Single Data Format Allowed. CLASS Calling Name Delivery Denied. CLASS Calling Name Multiple Data Format Allowed. Refer to the section “Configure CND Class of Service on CLASS sets” on page 1083 for details. ***Note:*** *These Class of Service entries are not valid for a Dial Intercom Group (DIG) set.*
CLS		
...		

LD 13 - Configure the CLASS modem unit (up to 255 CLASS modem units may be configured per system).

Note: While no new ISM limit is introduced by the CLASS feature, each CLASS modem unit TN is counted against the system TN limit and is reflected in the overlay banner.

Prompt	Response	Description
REQ	NEW CHG	Add new data Change existing data.
TYPE	CMOD	CLASS modem unit.
DMOD	1-127	Default Model number for this route (Small Systems)
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems ***Note:*** *Since the CLASS modem unit uses the octal-density Extended CLASS Modem Card (XCMC), up to 32 units can be configured on any XCMC card.*

Note: You can print the CMOD unit information using LD 20.

Feature operation

No specific operating procedures are required to use this feature.

CLASS: Visual Message Waiting Indicator

Contents

This section contains information on the following topics:

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Feature packaging 1121
Feature implementation 1122
Feature operation 1124

Feature description

The Custom Local Area Signaling Service (CLASS) Visual Message Waiting Indicator (VMWI) feature allows a CLASS set to receive a visual indication that messages are waiting. The visual indicator may be in the form of a lighting or flashing Light Emitting Diode (LED), or a special message on a liquid crystal display, or both. The type of visual indicator depends on the firmware of the CLASS set being used.

> ***Note:*** For a non-Class set, lighting the Message Waiting Lamp through a high voltage message (Voltage Message Waiting) is still supported and operates in the same way as it did before.

The visual indication message (an "ON/OFF" CLASS VMWI specific message) is delivered from the system to the CLASS sets using Frequency Shift Keying (FSK) signaling based on Bellcore specifications, using a CLASS modem (CMOD) unit. The CMOD units are configured using LD 13. Once configured, the CMOD units are shared throughout a multi-customer system. When an FSK message is presented to a CLASS set, an available CMOD unit is automatically allocated.

After all messages are retrieved, the visual indicator is cleared. The CLASS VMWI feature will not be able to turn off the message waiting indicator until the CLASS set has returned to the idle state, even though all of the messages have been retrieved by the user.

The CLASS VMWI feature supports the Bellcore Single Data Message Format (SDMF) and Multiple Data Message Format (MDMF) messaging. These formats are configured in the system database using LD 10. To use SDMF, a Class of Service of CNUS is required. For MDMF, a Class of Service of CNUA and/or CNAA is required.

The following description outlines the general functionality of the CLASS VMWI feature.

The following scenarios affect the sending of an ON or OFF message indicator to a CLASS set.

If the CLASS set is idle:

- the ON/OFF message is sent immediately to the set.

If the CLASS set is busy:

- as soon as the CLASS set becomes idle (goes on-hook), the system immediately sends the VMWI message to the set.

If the CLASS set or CMOD unit is disabled (using LD 32 or 30), or no CMOD unit is available:

- the CLASS VMWI feature will automatically abort the VMWI message. The CLASS set is then checked every 10 seconds, for up to 2 1/2 hours, until the set is detected to be idle. At that time, the system will immediately deliver the VMWI message.

If a CLASS set is in the middle of receiving a VMWI message, and the CLASS set goes off-hook:

- the VMWI message will be aborted. The CLASS set is checked every 10 seconds, for up to 2 1/2 hours, until the set is detected to be idle. At that time, the system will immediately deliver the VMWI message.

Note that, if the CLASS VMWI feature is waiting to send an "ON/OFF" message and another "ON/OFF" message is requested, only the last "ON/OFF" message is actually sent to the CLASS set.

The CLASS VMWI feature supports Nortel Networks and third party CLASS sets, if these sets are Bellcore VWMI compliant.

The following figures depict a typical feature operation and system resource allocation scenario for a CLASS Visual Message Waiting Indication being presented to a CLASS set. Note that the representation is for an ON message being delivered to the CLASS set. The same operation and system resource allocation applies for an OFF message being sent, except that the contents of the message would be different.

Figure 27
CLASS set is in idle state

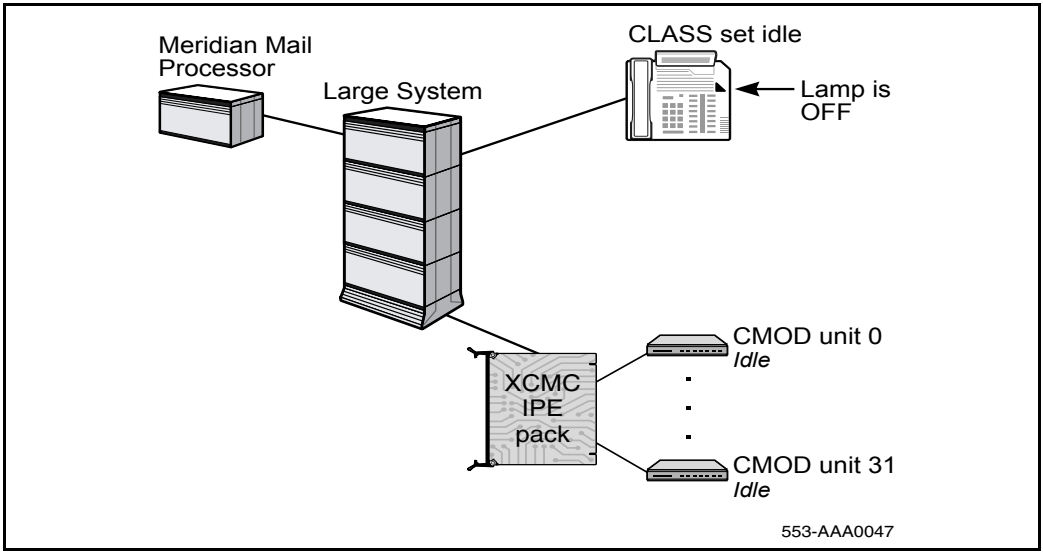


Figure 28
CLASS VMWI ON message in the process of being delivered to the CLASS set

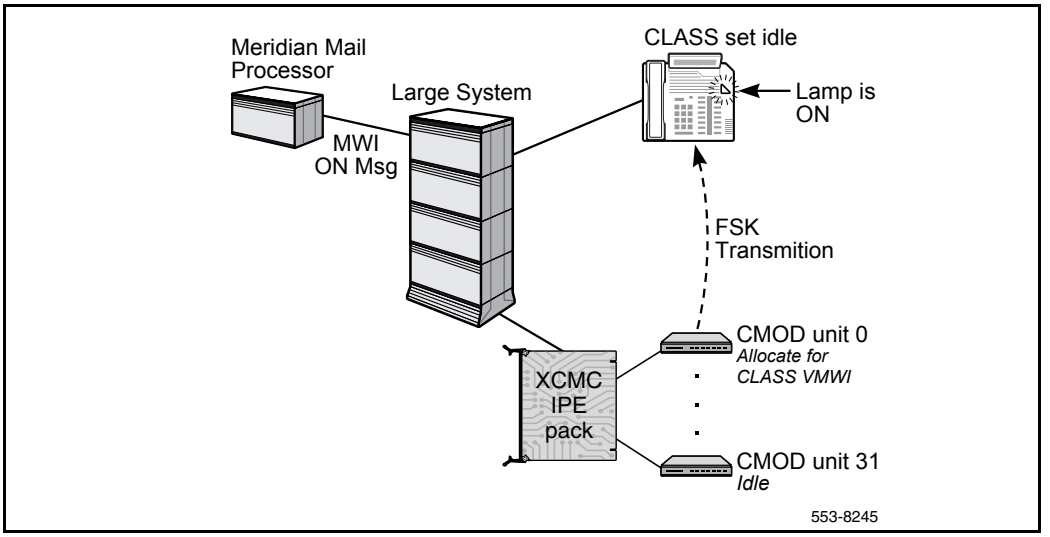
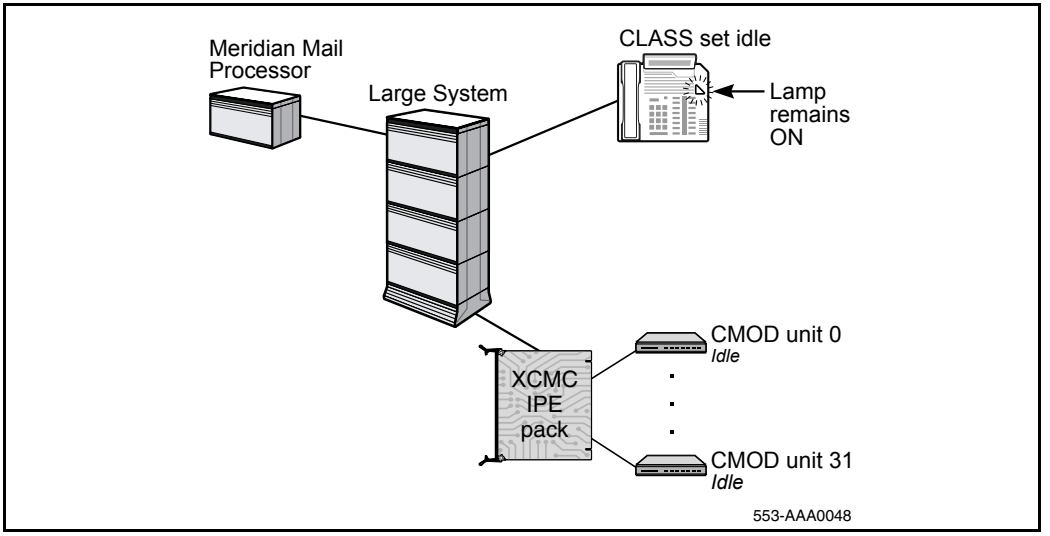


Figure 29
CLASS VMWI has been delivered to the CLASS set



Operating parameters

This feature has been developed for the North American residential market, as well as for the small business, hospitality, schools and universities, nursing homes and hospitals, and mobile home markets.

When a craftsperson uses LD 10 to administer a service change to a CLASS set, if the system software identifies, during overlay wrap-up, that the CLASS VMWI feature is waiting to send a visual indication to the set being serviced, a SCH1099 message will be generated, indicating that the VMWI pending message is lost and no retry will be attempted. The service change itself does not trigger any messages to the set.

Some CLASS sets, like the Nortel Networks's M9000 series of sets, support both voltage and CLASS (FSK) Message Waiting. However, once the CLASS set is configured in LD 10, then the system will no longer send voltage messages to the set.

If a CLASS set that previously used FSK messaging to receive Visual Message Waiting Indication has the Class of Service changed to be a non-CLASS set that uses Voltage Message Waiting, the visual indicator will not operate correctly. The following will occur:

- Voltage Message Waiting will work if the set is returned to the no message waiting state (visual indicator is dark) before switching over from FSK to voltage.
- Voltage message waiting will not work if the set is left at the message waiting state (visual indicator is lit) before switching from FSK to voltage.

It is up to the craftsperson to ensure that a CLASS set is correctly configured to support the CLASS VMWI feature.

Up to 255 CMOD units may be configured on a system. Once configured, the CMOD units are shared throughout a multi-customer system. When an FSK message is transmitted to a CLASS set, an available CMOD unit is automatically allocated.

If a CLASS set with a lit message waiting indicator retrieves all of its messages, the CLASS VMWI feature will not be able to turn off the message waiting indicator until the CLASS set has returned to the idle state.

Lamp audit does not apply to the CLASS VMWI feature.

There is a 2 1/2 hours time limit for a retry in the case of a pending message.

Up to 255 CLASS modem units can be configured on a system, and up to 32 CLASS modem units can be configured on any Extended CLASS Modem card.

Feature interactions

Attendant Console

When a CLASS set that is on a call with the attendant goes on-hook, the call will not be released until the attendant releases the call. The CLASS VMWI feature will wait until the call is actually disconnected before sending the VMWI message to the CLASS set that had already gone on-hook

Call Party Control Incoming

When a CLASS set on an active call is placed in Call Party Control Incoming (CPCI) state, the set will not be presented with any other incoming call. The CLASS VMWI feature will not send the VMWI message to the CLASS set to turn the visual indication ON or OFF, until the set releases the active call.

Make Set Busy

When a CLASS set is in the Make Set Busy state, the set will not be presented with incoming calls. Although no incoming calls are presented to the set, VMWI message can still be sent to the CLASS set to turn the visual indication ON or OFF, as long as the set is in the idle and on-hook state.

Meridian Mail

The CLASS VMWI feature makes use of Meridian Mail by having the Meridian Mail functionality communicate with the system to inform the CLASS set to turn the visual message waiting indicator ON or OFF. The system software determines the appropriate type of protocol based on the configuration of the CLASS set, and sends it to the set to turn the visual message waiting indicator ON or OFF.

Message Waiting

In order to support the CLASS VMWI functionality on a CLASS set, the Message Waiting feature has been enhanced to turn the message waiting indicator ON or OFF on the CLASS set through FSK signalling.

In addition, the CLASS VMWI feature can interwork with Stuttered Dial Tone. The Stuttered Dial Tone functionality (which is part of the Flexible Tones and Cadences feature) provides an audible indication, rather than a visual one, that a message is waiting. It is configured separately from the CLASS VMWI feature, using the Class of Service (CLS) prompt in LD 10. If the LPA (Lamp Allowed) response is entered, then a CLASS visual indication is sent. If the LPD (Lamp Denied) response is entered, then Stuttered Dial Tone is sent instead. Note that the Stuttered Dial Tone and CLASS visual indication are mutually exclusive. A CLASS set may receive either Visual or Stuttered Dial Tone as a message waiting indicator, but not both.

Permanent Hold

When a CLASS set is placed on permanent hold, the set is still treated as though it is presented with a phone call. The CLASS VMWI feature will not send the VMWI message to the CLASS set to turn the visual indication ON or OFF, until the set actually goes on-hook or the call is not presented to the set. At this time, VMWI message will be sent to the CLASS set to update the visual message waiting status.

Set Relocation

When a CLASS set is relocated, the CLASS VMWI feature will lose the message when the set finishes the relocation. An SCH1099 error message is printed to indicate that the CLASS VMWI message is lost and no automatic retry will be attempted.

Feature packaging

The following packages are required for the CLASS VMWI feature:

- Message Waiting Center (MWC) package 46
- CLASS Calling Number Delivery (CNUMB) package 332, or
- CLASS Calling Name Delivery (CNAME) package 333, and

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 10 - Configure a CLASS set as an analog (500/2500 type) set, and configure the required Class of Service for the set.
2. LD 13 - Configure the CLASS modem unit (up to 255 CLASS modem units may be configured per system).

LD 10 - Configure a CLASS set as an analog (500/2500 type) set, and configure the required Class of Service for the set.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		
DN	xxxx	Directory Number for the set
...		
CLS		Class of Service for the CLASS set
	MWA LPD	Message Waiting Allowed. Visual Indication (Lamp) Denied.
		Note: At least one of the following CLASS CLS must be allowed.
	CNUA CNUS CNAA	CLASS Calling Number Multiple Data Format Allowed. CLASS Calling Number Single Data Format Allowed. CLASS Calling Name Multiple Data Format Allowed.

...		

LD 13 - Configure the CLASS modem unit (up to 255 CLASS modem units may be configured per system).

Prompt	Response	Description
REQ	NEW CHG	Add new data Change existing data.
TYPE	CMOD	CLASS modem unit.
DMOD	1-127	Default Model number for this route
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems ***Note:*** *Since the CLASS modem unit uses the octal-density Extended CLASS Modem Card (XCMC), up to 32 units can be configured on any XCMC card.*

Note: You may print the CMOD unit information using LD 20.

Feature maintenance and diagnostics

LD 30 and LD 32

LD 30 and LD 32 handle CLASS VMWI messaging, in the case that a CMOD unit or a CLASS set is in the state of being disabled. Any CLASS VMWI message that is in progress will be aborted. A check every 10 seconds, for up to 2 1/2 hours, will be done on the CLASS set until another CMOD unit is found or the CLASS set is enabled, before a VMWI message is actually sent to the set to perform visual indication.

LD 77

LD 77 is used to monitor CLASS VMWI messaging.

Feature operation

No specific operating procedures are required to use this feature.

CLID on Analog Trunks for Hong Kong (A-CLID)

Contents

This section contains information on the following topics:

Feature description 1125
Operating parameters 1129
Feature interactions 1130
Feature packaging 1131
Feature implementation 1131
Feature operation 1132

Feature description

With the Calling Line Identification on Analog Trunks (A-CLID) feature and the DXUT-A card (NTRB37AA), on an incoming Central Office (CO) call, the system can extract information such as:

- Calling Party Number
- Calling Party Name
- Reason for absence of Calling Party Number or Name (if necessary)

The A-CLID information is treated similar to ISDN CLID for delivery to other modules and applications in the system, including the display on digital telephones and consoles at the local node and other network nodes (if any).

You can enable or disable A-CLID on an individual trunk port basis.

The A-CLID information passes to the terminating party, which includes:

- Trunks - ISDN (PRI/BRI/QSIG), R2MFC (DTI/DTI2, Analog)
  - — Calling Party Number information can be tandemed over all ISDN and R2MFC interfaces
  - — Calling Party Name information can be tandemed only on SL1 and QSIG ISDN interfaces. R2MFC does not support name information.
- Terminals - Attendant Consoles, Telephones (CLASS, 2208 with display, 2216, 2616, 2317, 5317, M3000, M3902, M3903, M3904, M3905)
- Applications - Call Pilot, Customer Controlled routing, Meridian Mail, Meridian Link, Symposium Call Center Server (calling party number only)

Figure 30 describes the feature operation.

Figure 30
System structure for CLID delivery

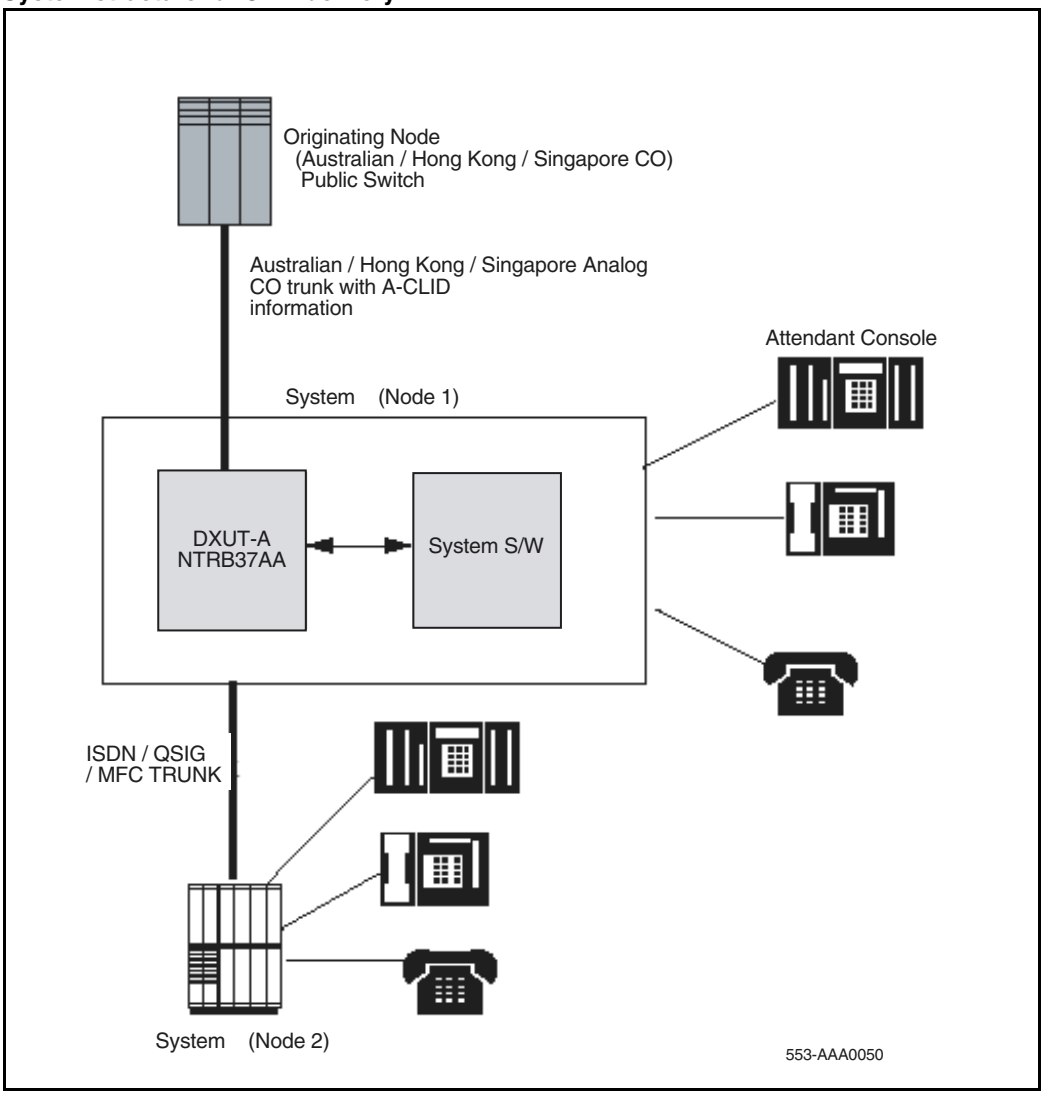


Table 44 shows the display format for different combinations of analog CLI information.

If you do not receive the calling number (“P” or “O” is received as the reason for absence), the display shows the P/O route access code number and member number.

If you do not receive the calling name (“P” or ”O” is received as the reason for absence), then the display shows “PRIVATE” or ”UNAVAILABLE”.

If you do not receive either the calling name, or its reason for absence, then the display shows “UNAVAILABLE”.

Table 44
Format of CLI information

Calling Number	Reason for Absence of Calling Number	Calling Name	Reason for Absence of Calling Name	Display Format
491893021	-	JOHN SMITH	-	JOHN SMITH 491893021
-	“O “	PAY PHONE	-	PAY PHONE O-8015-1
-	“O “	-	-	UNAVAILABLE O-8015-1
-	“O “	-	“O “	UNAVAILABLE O-8015-1
-	“P “	-	-	UNAVAILABLE P-8015-1
-	“P “	-	“P “	PRIVATE P-8015-1
P = Private	O = Other			

Operating parameters

The A-CLID feature is only for incoming loop start trunks (both supervised and non-supervised) in Hong Kong.

Direct Inward Dialing (DID) trunks do not support the A-CLID feature.

A-CLID requires one mode: On-hook data transmission with ring.

If you initialize the system while the CLID information is transferring from the DXUT-A card to software:

- you lose all CLID information not sent.
- you lose the un-established call (a call is established after the software receives the complete CLID information).

If the call is established, and you initialize the system, the call is restored. However, maintaining the displayed information is not guaranteed.

> ***Note:*** If system initialization occurs after the A-CLID call is in the agent queue, the ACD module must display the CLID information after the call is established.

If the cyclic redundancy check (CRC) fails, the call terminates without displaying the CLID ("UNAVAILABLE" and "O-Route access code number - Member number" for name and number display).

If data transmission stops half-way, the call terminates without displaying the CLID ("UNAVAILABLE" and "O-Route access code number - Member number" for name and number displays).

The system can receive, process, and display alphanumeric characters as CLI information. No CLI information displays if the CO transmits information in another format.

A maximum display of 20 digits and 27 characters is possible with this feature.

When the call is tandemed to an ISDN/MFC trunk, only the first 16 digits of A-CLID passes on.

This feature does not change the functionality of R2-MFC trunks. R2-MFC trunks display the route access code number when the CLI information is not available, and not the member number.

If the system receives unrecognized messages or parameter types, the call terminates without displaying the CLID ("UNAVAILABLE" and "O-Route access code number - Member number" for name and number displays).

The A-CLID feature delays the occurrence of the first ring. The terminating set rings when the complete FSK message is received, or the software times out waiting for the FSK message.

When a DISA call is abandoned, the CO trunks will be busy for a short period of time before disconnecting. This is characteristic of CO trunks.

Feature interactions

Attendant Call Extension

If an attendant extends a call from an incoming A-CLID CO trunk, the terminating set receives the analog CLI information.

CLASS: Calling Number and Name Delivery

The analog CLI information is given to the XCMC service pack to provide number and name display to analog CLASS telephones. The CLASS telephones can only display the CLI information when the incoming CO call is auto-terminated on that telephone.

For calls terminating on a CLASS set, only the:

- calling party number and name information display.
- first 15 characters of the information display.

Direct Inward System Access (DISA)

If a user connects to the system through Direct Inward System Access (DISA) dialing, the incoming trunk information is passed, not the DISA DN information.

Information Notification Service for Japan

The A-CLID feature does not work with the Information Notification Service for Japan (INS-J) feature.

Private Line Service

A-CLID information does display on the telephone.

Feature packaging

This feature requires Analog Calling Line Identification (ACLI) package 349.

Feature implementation

Use the Calling Line Identification Allowed (CLIA) Class of Service (CLS) in LD 14 to activate the A-CLID feature on an individual port basis.

Note: If EXUT is configured on a DXUT-A card pack, and the ACLI package is equipped, then the default CLS is Calling Line Identification Denied (CLID) for all incoming CO trunks to that card.

LD 14 – Configure analog CLI.

Prompt	Response	Description
REQ	CHG NEW x	Change existing data block. Add new data block to the system. Follow NEW with a value of 1-255 to create that number of consecutive trunks.
TYPE	COT	Central Office Trunk data block.
....	....	
XTRK	EXUT	Enhanced Extended CO trunk card.
....	....	
SIGL	LOP	Loop start
....	....	

CLS	 CLIA (CLID)	Class of Service options for trunks. Calling Line Identification Allowed Calling Line Identification Denied (default)

Feature operation

No specific operating procedures are required to use this feature.

Collect Call Blocking

Contents

This section contains information on the following topics:

Feature description 1133
Operating parameters 1135
Feature interactions 1137
Feature packaging 1140
Feature implementation 1140
Feature operation 1145

Feature description

In Brazil an automatic long distance collect call service called DDC is available. The Collect Call Blocking feature enables a system administrator to block DDC calls on incoming Direct Inward Dialing (DID) and Public Exchange/Central Office trunks (analog or DTI2). Under the following conditions, the system sends a special answer signal to the Central Office to indicate to the Central Office that collect calls cannot be accepted:

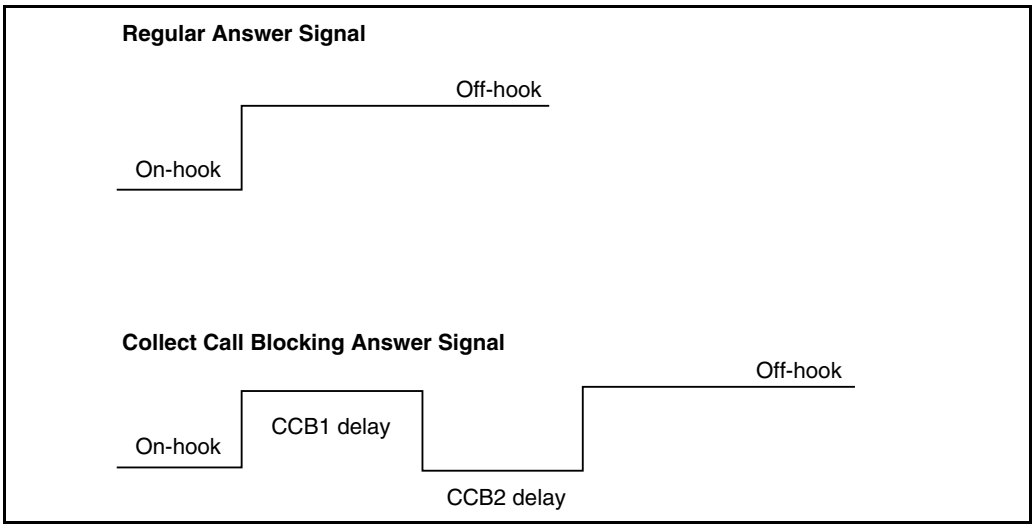
- The Collect Call Blocking (CCB) package 290 is enabled
- The incoming route has CCB enabled using the CCB prompt in the Route Data Block, and
- The call is answered by a CCB user (that is, Collect Call Blocking Allowed Class of Service or option).

Classes of Service and prompts have been introduced to inhibit specific users from receiving collect DID and Central Office calls. These can be configured for the following:

- analog (500/2500 type) telephones and Meridian 1 proprietary telephones through the Collect Call Blocking Allowed/Denied (CCBA/CCBD) Class of Service.
- Attendant and Network Alternate Route Selection calls on a per customer basis through CCBA/CCBD option.
- Automatic Call Distribution (ACD) queues through the CCBA prompt.
- Direct Inward System Access (DISA) through the CCBA prompt.
- Tandem calls dialed with Coordinated Dialing Plan (CDP) (Trunk Steering Code, Distant Steering Code) through the CCBA prompt.
- Tandem non-CDP calls through the CCBA prompt in the Route Data Block from the outgoing trunk route.

The system sends the CCB answer signal in place of the regular signal for incoming DID/CO calls from routes with CCB enabled, when a call is answered by a CCB user. If the call is a collect call, the CO will disconnect the call.

Figure 31
Collect Call Blocking answer signal compared to regular answer signal



Operating parameters

The Collect Call Blocking feature supports both analog and DTI2 trunks, and the following Intelligent Peripheral Equipment (IPE) cards:

- the NTCK16BB Extended Flexible COT Trunk Card (XFCOT) with firmware flash timing
- the NT8D14BA Enhanced Extended Universal Trunk Card (EXUT) containing the Centrex Switchhook Flash function in the firmware, and
- the NT8K14AK Extended Universal Trunk Card (XUT) which may be used if the Centrex Switchhook Flash is configured with software timing.

The Collect Call Blocking answer signal can only be sent in cases where answer supervision is provided by the system.

Once the modified answer signal is sent to the CO, the system has no control over how the call will be handled by the CO.

If a CCB user answers a call from a CO/DID route with Collect Call Blocking activated, the CCB answer signal is sent to the CO for all incoming DID and CO calls. For analog trunks, the user will experience clicking on the line and a temporary break is speechpath (0.5 to 2.5 seconds) while the CCB answer signal is being sent.

If the XFCOT and EXUT cards do not have flexible firmware timing, the CCB flash portion of the CCB answer signal will be ignored by firmware, and the regular answer signal will be returned to the CO. However, software controlled signaling can be done with EXUT cards.

In a standalone environment, all input from a set (except from the Release key) is ignored while the Collect Call Blocking answer signal is being sent.

Collect Call Blocking is applied to attendants on a customer basis only; it cannot be applied on a tenant basis.

The answer signal returned for a call from a route with CCB enabled and that is Network Attendant Service (NAS) routed is determined by the customer option on the source node. Thus, NAS routing can be configured across any Meridian Customer Defined Network environment, but the source node determines the answer supervision sent to the CO.

Call Detail Recording (CDR) record timing begins on the first answer of the CCB answer sequence. For this reason, CDR records will be generated for incoming calls to CCB users across routes on which CCB is enabled. If the call is collect, and is dropped, a CDR record of approximately CCB1 + CCB2 length will be generated.

For data calls, all calls will be answered with the CCB answer signal, if CCB is enabled. This may have an effect on data protocols, while CCB signaling is taking place.

If firmware timing is used (FWTM = YES in Overlay14) for sending the CCB flash, the CCB2 timer is downloaded to the card before sending the firmware flash. If the CCB2 timer is changed in the Route Data Block, either the card has to be enabled or the switch has to be initialized to get the new CCB2 timer downloaded to the card.

Feature interactions

Automatic Answerback

The Automatic Answerback (AAB) feature, when assigned to a Meridian 1 proprietary telephone, allows any incoming call to a single-appearance Prime Directory Number (PDN) to be answered automatically. If an incoming DID or CO call terminates on a set with the AAB feature enabled, the call is automatically answered after one ring. If the set has a CCBA Class of Service, the CCB answer signal is provided in the place of the regular answer signal.

Automatic Call Distribution

Collect Call Blocking can be enabled on an ACD queue basis. Hence, if an incoming CO or DID call is answered by an ACD agent, the answer supervision signal that is returned to the CO is determined by the value of the CCBA prompt in LD 23. While the CCB answer signal is being sent, the same limitations apply to ACD as apply to sets with CCBA Class of Service.

Automatic Call Distribution Interflow

If an ACD call from a route with CCB enabled is diverted to an interflow DN, and answer supervision has not already been provided, the answer signal returned to the CO depends on the source ACD queue. The CCB answer signal is returned to the CO if the source ACD queue has CCB enabled.

Automatic Call Distribution Night Call Forward

During Night Call Forward (ACD) call processing, the source ACD queue is removed or overwritten. Therefore, the CCB treatment given will be based on the queue the call happens to be in at the time the call is answered.

Automatic Call Distribution Night RAN Route Announcement

If an ACD call from a route with CCB enabled is diverted to a Night RAN route (defined by NRRT in the ACD block), the CCB signal returned to the CO depends on the source ACD queue. If the source ACD queue has CCB enabled, the CCB answer signal is sent to the CO.

Automatic Call Distribution Overflow

If an ACD call from a route with CCB enabled is diverted to an overflow ACD DN and answer supervision has not already been provided, the answer signal returned to the CO depends on the source ACD queue from where the call came. If the source ACD queue has CCB enabled, the CCB signal is sent to the CO.

Autoterminate

If an incoming DID or CO call from an autoterminate trunk terminates on a set or ACD queue with a CCBA Class of Service, the CCB answer signal is provided in place of the regular answer signal.

Basic Rate Interface (BRI) Sets

For BRI sets CCBA/CCBD Class of Service cannot be programmed. Therefore, it is not possible to prevent BRI sets from accepting DDC collect calls.

Central Answering Position

The answer signal returned to the CO for calls that get answered by a Central Answering Position (CAP) is determined by the source ACD configuration and not the customer option (CCBA/CCBD in LD 15) on the source node.

Centralized Attendant Service

The answer signal returned to the CO for calls that get answered by a Centralized Attendant Service is determined by the customer option (CCBA/CCDB in LD 15) on the source node.

Centrex Switchhook Flash

A Centrex Switchhook Flash cannot be invoked by another feature while the CCB answer signal is being sent.

Malicious Call Trace - Enhanced

If a station activates Malicious Call Trace (MCT) while the CCB answer signal is being sent, MCT activation is ignored. This also applies to the case when MCT is activated from a remote node.

Meridian Mail

Because Meridian Mail is configured using ACD queues, the same interactions exist as in the ACD case. When Meridian Mail sends a call answer message to the system, the CCB configuration in the source ACD queue is used to determine if a CCB answer signal should be sent to the Central Office. All mail boxes using the same ACD queue to access Meridian Mail will get the same CCB treatment.

If some of the mail boxes are allowed to receive collect calls, this may be a problem. A possible solution is to configure two ACD queues on the system to access Meridian Mail. One queue would have collect calls allowed and the second queue would have collect calls denied.

Network Automatic Call Distribution

The answer signal returned to the CO for a network ACD call from a route with CCB enabled is determined by the source ACD queue. If the source ACD queue has CCB enabled, the CCB answer signal is returned in place of the regular answer signal.

Pilot DN

If an incoming DID or CO call has CCB enabled and is routed to a pilot DN, the answer signal returned to the CO is determined by the CCB configuration of the terminating station.

Private Line Service

If an incoming DID or CO call from a private line trunk terminates on a set with a CCBA Class of Service, the CCB answer signal is provided in place of the regular answer signal.

Recorded Announcement

A Recorded Announcement (RAN) route is defined as having CCBA YES or NO, which is used if Coordinated Dialing Plan (CDP) or ACD queues were not used to get to the RAN route. If the call is routed through ACD/CDP to terminate on RAN, the CCB treatment will depend upon the CCB data of the ACD/CDP, and not of the RAN route.

Tandem to Unsupervised Trunk

If an incoming DID or CO call tandems to an unsupervised trunk before it terminates, the answer signal is sent by time-out. Therefore, any CCB tandem calls made to unsupervised trunks will not have the CCB answer signal sent until the time-out occurs.

Trunk Hook Flash

If a station activates Trunk Hook Flash (THF) while the CCB answer signal is being sent, THF activation is ignored.

Feature packaging

Collect Call Blocking (CCB) package 290 must be provisioned to activate this feature.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Enable Collect Call Blocking on a route and configure timers.

2 LD 14 – Setup the firmware timing for XFCOT and EXUT cards.

3 LD 15 – Allow Collect Call Blocking for attendants.

4 LD 10 – Enable Collect Call Blocking for analog (500/2500 type) telephones.

5 LD 11 – Enable Collect Call Blocking for Meridian 1 proprietary telephones.

6 LD 23 – Enable Collect Call Blocking on ACD queues.

7 LD 24 – Enable Collect Call Blocking on DISA blocks.

8 LD 87 – Enable Collect Call Blocking on CDP Steering codes.

LD 16 – Enable Collect Call Blocking on a route and configure timers.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	aaa	Trunk type. Must be COT, DID, FEX, or WAT for CCB.
...		
M911_ANI	NO	M911 route. Must be set to NO to enable CCB.
...		
ISDN	NO	ISDN route. Must be set to NO to enable CCB.
...		
ICOG	IAO ICT OGT	Incoming and outgoing, incoming, or outgoing. Must be either IAO or ICT to enable CCB. Must be either IAO or OGT to get the CCBA prompt for outgoing calls.
...		
CNTL	(NO) YES	Change to controls or to timers.
...		
CCB	(NO) YES	Collect Call Blocking enabled or disabled on incoming route. CCB package 290 is required. Enter YES to obtain CCB timer prompts.
CCB1	512-(1536)-4992	Collect Call Blocking delay timer 1 in milliseconds. Input rounded to the next multiple of 128 milliseconds.

CCB2	500-(1520)-2550	Collect Call Blocking delay timer 2 in milliseconds. Input rounded to the next multiple of 10 milliseconds. If any CCB route members (trunks) are using firmware timing (FWTM = YES in LD 14), changes to the CCB2 timer value will not take effect until the new timer value is downloaded to the card. This can be done by enabling the card or initializing the switch.
CCBA	(NO) YES	Collect Call Blocking allowed or denied for outgoing route.

LD 14 – Setup the firmware timing for XFCOT and EXUT cards.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add. Change.
TYPE	DID COT FEX WAT	Trunk Type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	EXUT XCOT	Type of card.
FWTM	(NO) YES	Firmware timing for flash. Enter YES to enable firmware timing.
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
SUPN	YES	Answer supervision required.

LD 15 – Allow Collect Call Blocking for attendants.

Prompt	Response	Description
REQ:	NEW CHG	Add. Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	CCBA	Allow Collect Call Blocking. CCBD =Default

LD 10 – Enable Collect Call Blocking for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add. Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	(CCBD) CCBA	(Deny) allow Collect Call Blocking.

LD 11 – Enable Collect Call Blocking for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add. Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	(CCBD) CCBA	(Deny) allow Collect Call Blocking.

LD 23 – Enable Collect Call Blocking on ACD queues.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	ACD	ACD data block.
CUST	xx	Customer number, as defined in LD 15
ACDN	xxxx	ACD Directory Number.
...		
CCBA	(NO) YES	(Deny) allow Collect Call Blocking.

LD 24 – Enable Collect Call Blocking on DISA blocks.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	DIS	DISA data block.
CUST	xx	Customer number, as defined in LD 15
...		
DN	xxxxxxx	DISA Directory Number.
...		
CCBA	(NO) YES	(Deny) allow CCB answer signal to be sent.

LD 87 – Enable Collect Call Blocking on CDP Steering codes.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
CUST	xx	Customer number, as defined in LD 15
FEAT	CDP	Coordinated Dialing Plan.
TYPE	TSC DSC	Steering code type.
...		
CCBA	(NO) YES	(Deny) allow Collect Call Blocking.

Feature operation

No specific operating procedures are required to use this feature.

Conference Warning Tone Enhancement

Contents

This section contains information on the following topics:

Feature description 1147
Operating parameters 1148
Feature Interactions 1148
Feature packaging 1148
Feature implementation 1149
Feature operation 1150

Feature description

The Conference Warning Tone Enhancement feature was developed to meet the Italian requirements to distinguish between a conference warning tone and a true intrusion. A Conference Warning Tone warns users that they are involved in a conference connection. An Intrusion Tone warns users involved in a conference of an intrusion into their connection.

Prior to this development, the Intrusion Tone was also used as a Conference Warning Tone, and the two tones could only be distinguished by their cadences, not by their tone frequency. With this feature, a separate Conference Warning Tone can be defined, with its tone and cadence defined in overlay program. This tone and cadence can only be programmed in Flexible Tones and Cadences (FTC) table 0.

The Conference Warning Tone can be enabled or disabled using the CWFT prompt in overlay program. When the tone is enabled, a permanent speech path connection is placed from the tone circuit to the conference circuit to provide the tone to all parties connected in a conference. The tone and cadence for the Conference Warning Tone can be programmed to be distinctively different from the tone and cadence produced by the Intrusion Tone given by such features as Barge-In, Break-In, or Busy Verify.

Operating parameters

This feature only applies to the Succession 1000M Cabinet and Meridian 1 Option 11C Cabinet.

The Conference Warning Tone will only be contained and defined within FTC Table 0. All other FTC tables will not contain any references to the Conference Warning Tone and this tone cannot be copied to any other FTC table by a numbered response to the DFLT (Default to existing FTC tone table) prompt in overlay program. Since existing code does not allow FTC table 0 to be deleted, there is no danger of deleting the Conference Warning Tone.

For cadence tables, table numbers above 15 should be chosen for the Conference Warning Tone, because cadence tables numbers below 15 are influenced by Software Controlled table numbers.

Feature Interactions

Tones and Cadences

There are no changes to the limitations to cadence numbers entry values. The same restriction still applies.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 56 – Configure Conference Warning Tone.

2 LD 97 – Enable Conference Warning Tone.

LD 56 – Configure Conference Warning Tone.

Prompt	Response	Description
REQ	CHG	Modify existing data.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0	FTC table number 0. Only table number 0 can be used to make changes to the Conference Warning Tone.
...		
RING	NO	Modify the ringing feature definitions.
HCCT	YES	Modification of the hardware controlled cadence tone definitions allowed.
...		
- EEST	NO	End-to-end Signaling type.
- CFWT		Conference Warning Tone.
- - XTON	0-(3)-255	The Conference Warning Tone number. A tone number provided by the tone circuit.
- - XCAD	0-(19)-255	The Conference Warning Tone cadence number. Cadence number must have been previously set up within LD 56 by responding to the TYPE prompt with FCAD.

LD 97 – Enable Conference Warning Tone.

Prompt	Response	Description
REQ	CHG	Modify existing data.
TYPE	XCTP	Conference/TDS/MF Sender card parameters.
CPAD	 (0) 1	Conference pad values. Use software pad values. Use pad values defined by switch settings.
DTMF	0-(14)-255	Tone table of the first Dual-tone Multifrequency digit to be used.
CFWT	(NO) YES	(Disable) enable Conference Warning Tone.

Feature operation

No specific operating procedures are required to use this feature.

Conference

Contents

This section contains information on the following topics:

Feature description 1151
Operating parameters 1152
Feature interactions 1153
Feature packaging 1165
Feature implementation 1166
Feature operation 1167

Feature description

Conference adds additional parties to an established call. The maximum is three or six, depending on the Conference feature assigned to the conference call originator. Not all conference parties have to be local to the system, although one party must be an internal Directory Number (DN) to uphold the conference connection. The attendant can also establish six-party conferences.

Meridian 1 proprietary telephones require a separate Conference 3 or Conference 6 key/lamp pair. M2317 and M3000 Touchphones establish conference calls by means of a softkey. Analog (500/2500 type) telephones use the switchhook to establish a three-party conference.

The six-party Conference (C6A) Class of Service enables analog (500/2500 type) telephones to establish a six-party conference, which operates the same as a three-party conference, with the exception of Conference Control operation.

Conference Control disconnects an unwanted third party (trunk only) from a three-party conference. Analog (500/2500 type) telephone users implement this feature by means of switchhook flash. Telephones with the six-party conference capability implement Conference Control by dialing SPRE + 87.

It is recommended that all analog (500/2500 type) telephones have either the three-party conference (C6D) Class of Service or the six-party Conference (C6A) Class of Service to avoid confusion when using Conference Control.

Operating parameters

Due to the possibility of getting annoying noise (squealing, for example) when two or more trunks are involved in a conference, it is strongly recommended that no more that two trunks be involved in the process.

At least one party in the conference must be a telephone on the local system for the duration of the conference call.

Attendant Administration does not support the implementation of six-party conference for analog (500/2500 type) telephones. An error message is displayed if an attempt is made to remove Transfer Allowed (XFA) Class of Service for analog (500/2500 type) telephones with a C6A Class of Service.

A Transfer allowed (XFA) Class of Service is required for a three-party conference (C6D) and is also a prerequisite for the six-party conference Class of Service (C6A) on analog (500/2500 type) telephones.

Dial access of Conference Control is provided only for analog (500/2500 type) telephones with a C6A Class of Service.

The number of timeslots is limited to 30 per conference loop. For Small Systems, a maximum of five simultaneous conferences, each consisting of six conference users, is supported per conference loop.

Double conferences are blocked to avoid conference chains (that is, set A and set B are on an established call. The user of set B presses the Conference key to call set C. Any attempt by set C to initiate another conference is blocked if set B has not yet completed the conference).

While the originating side of a call is linked to a transfer or conference key (that is, the originator of a transfer/conference call has not yet completed the transfer/conference), the terminating side cannot initiate a transfer or conference. Conference calls cannot be transferred.

A warning tone is available for conference calls. When the option is enabled, the tone lets callers know that they are entering a conference call. The switch for this option is preset to disable the warning tone. For information on the switch settings for the NT8D17 Conference/TDS card, refer to *Circuit Card: Description and Installation* (553-3001-211).

Feature interactions

500/2500 Line Disconnect

If one of the parties in the conference is connected to a 500/2500 port that is in turn connected to a Voice Response Unit (VRU), dial tone is provided to the 500/2500 port when all the other parties in the conference disconnect. This feature enhancement applies in the same way to Call Transfer and Hunting.

AC15 Recall: Transfer from Meridian 1

The use of the Conference key does not activate the AC15 Recall: Transfer from Meridian 1 feature. Conference call is not supported because it is not possible to have two parties on the same trunk.

AC15 Recall: Timed Reminder Recall

The conference feature is sometimes used to perform a transfer when a controlling party establishes a call, the controlling party establishes a conference with a third party and releases, and a call is established between the two remaining parties.

If an established call is extended over a trunk to initiate a conference call, this conference call cannot be set up if this trunk has answer supervision and the called extension has not answered. The AC15 Timed Reminder Recall feature cannot be activated by using the conference feature to extend a call over an AC15 TIE trunk, because the AC15 TIE trunk must have answer supervision and the called extension must be ringing.

AC15 Recall: Transfer from Norstar

It is not possible in any situation with Transfer from Norstar to establish a three-party conference. It is not possible for an AC15 trunk to initiate a consultation if it is involved in a conference.

Advice of Charge for EuroISDN

If a set is participating in a conference, no charge is displayed for that set. Whenever an ISDN CO trunks that provides Advice of Charge (AOC) is added to a conference, the call charging information, received from the network, accumulates against the set that initiates the call.

Once the last set involved in a conference call disconnects, a search is made of all trunks remaining in the conference call to determine which trunk has been established in the call for the longest period of time. This trunk becomes the chargeable TN. Once this trunk disconnects, the process is repeated so a new chargeable TN can be located.

Attendant Barge-In
Attendant Busy Verify

Conference Control cannot be activated if an attendant has used Barge-In or Busy Verify during a conference that involves a trunk.

Attendant Break-In

If the attendant cannot break in to a conference call because the call is supporting the maximum number of callers, busy tone continues and the Break-In key lamp flashes.

Attendant Console

Three-party Conference (C6D) allows analog (500/2500 type) telephones on established calls to flash the switchhook and Dial 0 to talk to the attendant. Six-party conference users follow the same sequence, but the conference loop is seized and the call is treated as a conference call. When only two parties remain from the conference, the call is returned to a simple call if neither of the remaining parties is an Attendant Console.

Attendant Overflow Position

An Attendant Overflow Position (AOP) call answered on an AOP DN may be conferenced with another DN.

Autodial Tandem Transfer

The Autodial Tandem Transfer feature is blocked during Conference and No Hold Conference calls.

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Conference (A03 or A06) key is ignored.

B34 Codec Static Loss Plan Downloading
B34 Dynamic Loss Switching

When a conference connection is established, no pads are switched in on the trunk side; any extra loss that is required is provided by the conference circuit based on an algorithm which takes into account the number of lines and trunks.

Call Forward All Calls

On analog (500/2500 type) telephones, Call Forward All Calls can be activated or canceled during a conference call.

Call Forward by Call Type

Calls modified by Conference receive Call Forward by Call Type treatment for the conferenced telephone. If party A calls party B, and B tries to conference in party C, the forwarding DN and Class of Service are that of C. For example, Joan and Bob are in conversation, and they try to conference in Mack. Mack is not at his desk, so the attempted conference call is sent to the destination associated with Mack's telephone.

Call Page Network Wide

A station set or Attendant Console that conferences an external Call Page Network Wide (PAGENET) uncontrolled call is not blocked. However, an external PAGENET controlled call is blocked.

Call Park

A parked call can be accessed after Conference is activated

Call Party Name Display

When pressed during an active call, or to set up a conference, the Conference, Connect, or Join Parties key clears the display. The telephones involved in the conference have blank displays. If the conference returns to a two-way only call, each telephone displays the DN and name of the other telephone.

Call Pickup

This feature cannot be activated during a conference call. Meridian 1 proprietary telephones can activate Call Pickup if an idle Directory Number (DN) key is available. The conference call must be put on hold before pressing the idle DN key to pick up the call.

Call Pickup Network Wide

Call Pickup Network Wide may be used to pickup an enquiry call from a conference, subject to the same limitations as apply to Call Transfer.

Call Transfer

Conference can be used to transfer calls, eliminating the need for a separate Call Transfer key/lamp pair on Meridian 1 proprietary telephones. Calls in the ringing state cannot be transferred with Conference. The third party must answer before the transfer can be completed.

A conference can also be established after initiating a Call Transfer operation. After the third party answers, pressing the Conference key establishes a three-way conference.

When a switchhook flash transfers calls on analog (500/2500 type) telephones with three-party conference (C6A) Class of Service, the transferring party goes on hook, leaving the other two parties established. Telephones with a C6A Class of Service involved in a conference having more than three parties must add the last party to the conference, then flash the switchhook and go on hook to complete the transfer.

Called Party Disconnect Control

Trunks with Called Party Disconnect Control allowed are treated as trunks without disconnect supervision when conferenced.

Calling Party Name Display Denied

Call Party Name Display, and thus the Calling Party Name Display Denied enhancement, do not apply to conference calls.

Calling Party Privacy

The Calling Party Privacy (CPP) feature will pass the Privacy Indicator to the terminating set to inhibit the display of the Calling Party Name and Number if the Conference feature is used for the purpose of performing a transfer.

Camp-On, Forced

Telephones involved in Conference calls cannot be camped on to. Overflow tone is returned to telephones attempting Forced Camp-On.

Centrex Switchhook Flash

Centrex Switchhook Flash (THF) allows conference calls through the CO. It can be invoked only if there is an established call connected to an outside trunk. If the telephone is engaged in internal conference calls, THF cannot be used.

Charge Account and Calling Party Name

Conference calls produce multiple Call Detail Recording (CDR) records. Whenever a new trunk is added to a conference, the connection between the connected telephone and the trunk is recorded, and a connection to the conference loop is established. This causes CDR to generate a start record with the telephone and trunk identified as the involved parties. As trunks are removed from a conference, CDR end records are produced. These records may identify different telephones or conferences as the local parties.

China – Attendant Monitor

If any party involved in a monitored call attempts to activate conference, monitoring is immediately deactivated. With Attendant Monitor active, the attendant cannot create a conference without first disabling the Attendant Monitor feature.

China – Supervised Analog Lines

If a terminal device answers an incoming call and then initiates a conference, no battery reversal answer supervision signal is extended to the terminal device when new parties of the conference answer. However, a hook flash disconnect supervision signal is extended to the terminal device when the last party in the conference disconnects.

If a terminal device initiates a conference, battery reversal answer supervision is extended to the terminal device when the first party answers. No polarity change is made when additional parties are added to the conference. The polarity is reverted to normal when the terminal device disconnects or when the last party in the conference disconnects.

China – Toll Call Loss Plan

Toll Loss Plan is not supported when a conference is in progress. When a local party connecting to a toll call makes a conference call, the pad levels on the ONS line card are switched back to their original (non-toll call) values. Then, the existing Conference algorithm takes care of the necessary pad switching. This would not alter the existing conference call in terms of loss levels.

When a conference call joins in a toll call, the Toll Loss Plan is not effective.

When a conference call involving a toll call becomes a two-party call, the Toll Loss Plan is applied on the set and DTI2 trunk.

The conference pad switching algorithm is not changed for the Toll Loss Plan, since the 7 db requirement does not apply to a Conference call.

Controlled Class of Service

If Controlled Class of Service (CCOS) is activated at a telephone involved in a conference call, established Central Office or toll calls are not affected. The CCOS restriction level is applied immediately, and no new calls can be initiated from the conference. The telephone remains in the CCOS active state after the conference is terminated.

Controlled Class of Service, Enhanced

If Controlled Class of Service (CCOS) is activated at a telephone on a conference call, established Public Exchange/Central Office or toll calls are not affected. The CCOS restriction level is applied immediately; however, no new calls can be initiated from the conference. That telephone remains in the CCOS state after the end of the conference.

Dial Access to Group Calls

The Conference feature cannot be applied to a Group Call.

Dial Intercom

If an analog (500/2500 type) telephone is part of a Dial Intercom Group (DIG), the user of the telephone can conference only with another user whose telephone is within the same Dial Intercom Group (DIG).

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is denied if the requested party is established in a local conference, or if the requested party is involved in an enquiry call. These restrictions may apply to the unrequested party depending on the connection being used between the requested and unrequested parties.

End-to-End Signaling

The Attendant Console and the telephone receiving Attendant End-to-End Signaling cannot both activate End-to-End Signaling simultaneously.

Group Call

Neither Call Transfer nor Conference can be initiated during a Group Call. If an analog (500/2500 type) telephone user flashes the switchhook during an established Group Call, the user is dropped from the call.

Held Call Clearing

Active Conference calls are cleared by an on-hook or Release key action. Conference calls being held are cleared by an on-hook action only, and not by a Release key action. In either case, all other parties on the conference remain connected.

Hot Line

A Flexible Hot Line (non-enhanced) telephone cannot place conference calls, but an Enhanced Hot Line telephone can activate the conference feature. If the Hot Line restriction option is set, the conference call can terminate only to other Hot Line telephones. If the restriction option is not set, the conference call can terminate to any type of telephone.

ISDN QSIG/EuroISDN Call Completion

A Call Completion request cannot be made on a conference call attempt.

ISDN QSIG Name Display

An incoming QSIG call with name display presentation allowed is conferenced locally. When a conferee drops out of the conference, calling party's name information is displayed and is passed on to another conferee. Name display information remains until the last local set remains on the call.With presentation restricted, the calling party's name information is not displayed as conferees leave the call.

In-Band Automatic Number Identification

If an agent activates the Conference feature while active on an In-Band Automatic Number Identification (IANI) call, the display is cleared. The display remains clear while the Conference call is active. If the conferenced party releases first, the ANI number appears on the agent's display.

Malicious Call Trace

When a station or console that is on the conference loop activates the MCT feature, the trace record shows only the conference loop number and conference number as the ORIGTN, and the Terminal Number (TN) of the station or console that activated the feature as the TERTN. No information on the other parties in the conference is given.

Malicious Call Trace - Enhanced

If MCT is activated during a conference, the trace record shows the conference number and the conference loop number. Trace records are printed for each party involved in the conference. The originator of the call's trace record is printed first.

Meridian Mail Conference Control

Three- and six-party conference allows 2500 telephones to disconnect from Meridian Mail by dial access during a conference call.

A 2500 telephone on an established call flashes the switchhook to place the existing call on Consultation Hold. After receiving special dial tone, the user dials the third party. If the third party does not answer, the call is forwarded to Meridian Mail. If the 2500 telephone flashes the switchhook again, a three-party conference is established, including Meridian Mail. If the user does not flash the switchhook at this time, Privacy is in effect and the user can disconnect from Meridian Mail by dial access before returning to the original call. This can be done if the user is in conference or on a simple two-party call.

To disconnect from Meridian Mail, press octothorpe (#) to stop the recorded greeting, octothorpe (#) to stop recording your message, and 83 to disconnect. To disconnect from any other message system connected to the system, press 3 to stop the recorded message and the asterisk (*) to disconnect.

Meridian 911

When a call is answered, and then conferenced, the trunk priority is lost (the conference consultation call is an internal call and treated as low priority by the software). This operation is the same for normal calls and 911 calls.

Meridian 911 - Call Abandon

M911 abandoned calls cannot be conferenced.

Message Registration

The party that originates a call is charged. The charge cannot be moved to another party using Conference.

Multi-Party Operations

Current Conference feature for analog (500/2500 type) telephones with C6A is not affected by conference with TSA Class of Service.

Multi-Party Operations – Call Join

The Call Join feature allows a user of a system or digital telephone to conference in or transfer a third party to a party held on the user's telephone, without having to dial the third party. The user can then hang up.

Multi-Party Operations -Three-Party Service
Multi-Party Operations Enhancements

The patience tone or the Misoperation ringback is not applied to a conference party.

Music

With basic Music on Hold, when a call is placed on consultation hold while a Conference is being established, music does not play. Enhanced Music (EMUS) package 119 is required for music on consultation hold. See the "Music, Enhanced" feature description in *Features and Services* (553-3001-306), Book 2.

Music, Enhanced

The held party receives Music when the Conference key is pressed, while the conference is being established, and whenever the conference is reduced to two parties with one party on Hold. Once the conference is established, Music is no longer provided.

A Six-party Conference operates the same as a Three-party Conference.

Network and Executive Distinctive Ringing

If a new party is to be included in an established conference, the ringing that is applied to the set of the new party depends on the sets of the established parties. The system scans the trunks and sets of the conferees for a trunk marked as distinctive or a set designated as executive. The ringing cadence of the new set depends on the highest index found by the scan.

Network Intercom

A Conference call may involve a mixture of intercom and regular DN keys.

No Hold Conference

This feature can be enabled at any time that a regular Conference-6 feature can be activated.

Off-Hook Alarm Security

The Off-Hook Alarm Security (OHAS) line lockout treatment occurs when a telephone associated with an OHAS DN initiates a Conference call and the ASTM expires. Only the Conference initiator receives the OHAS treatment; other conferees remain in Conference. If the initiator of the Conference call presses the Conference key, the OHAS DN is conferenced in with the other conferees.

On Hold on Loudspeaker

It will not be possible to conference the loudspeaker call to another party.

Override

A conference call cannot be entered by using Override.

Override, Enhanced

Telephones involved in conference calls cannot be force camped on or Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Paging

Paging trunks cannot be conferenced.

Periodic Pulse Metering (PPM)

Whenever a PPM trunk is added to a conference, a CDR Start record is generated, if CDR is equipped on the trunk. The PPM pulse counts from the trunk are accumulated against the party who initiated the call. If a party who adds a PPM trunk to the conference disconnects while the conference is still in progress, read requests are sent to the PPM trunk to read the residual count. Then, the on-board counter is cleared, the residual count is added to the temporary meter, and the contents of the temporary meter are added to the terminal meter. A CDR Transfer (X) record is then printed against this party, and the temporary meter is cleared. The party that is charged is the one that has been in conference the longest. When a trunk with disconnect supervision disconnects, a CDR End record is immediately printed. For trunks that do not provide a disconnect signal, their CDR records are not printed until the last party disconnects from the conference.

Privacy Override

The Conference feature can be used to add other parties to a Privacy Override connection.

Recorded Announcement Trunk

A Recorded Announcement (RAN) trunk cannot be Conferenced.

Recorded Telephone Dictation

Dictation trunks cannot be conferenced.

Ring Again

This feature cannot be activated during a conference call.

Station Activity Records

For a set with Class of Service Call Detail Monitoring Allowed (CDMA) involved in a call with a trunk, a Station Activity Record is produced only when that set conferences in the first party. Conferencing of all subsequent parties does not generate a “D” record. An additional “D” record is produced when the last conferee with Class of Service CDMA connected to the trunk goes on hook. This does not affect any other CDR record generation during a conference.

Trunk Access from any Station

A switchhook flash on analog (500/2500 type) telephones results in special dial tone. Dialing SPRE + 4 (TAFAS access code) then picks up an incoming Trunk Access from any Station (TAFAS) call. A second switchhook flash reconnects the user to the original conference call. The call picked up by TAFAS is put on Consultation Hold. No other action can be taken with a call picked up in this way during an established conference call.

Trunk Barring

The originator of a conference call can connect only to a barred route on a consultation basis. A switchhook flash from an analog (500/2500 type) telephone results in a reestablished connection with the Originating Trunk Connection. The user of a Meridian 1 proprietary telephone must release the barred connection to return to the Originating Trunk connection, or the conference containing the Originating Trunk connection; operating the Conference key on a Meridian 1 proprietary telephone has no effect. An attendant may return to the Originating Trunk Connection, or the conference containing the Originating Trunk Connection, by releasing the barred connection. This is done by pressing the RLS DEST key; pressing the Conference key has no effect.

Trunk to Trunk Connection

Trunk to Trunk Connection allows external trunks to remain established in a call, provided that all external trunks involved have disconnect supervision. With respect to charging costs associated with a conference call, once the last set involved in the conference call disconnects, a search is made of all remaining trunks in the call to determine which call is established in the call for the longest period of time. This trunk is the chargeable Terminal Number (TN). This process is repeated to find the next chargeable TN.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable Conference 3 or Conference 6 for analog (500/2500 type) telephones.

2 LD 11 – Enable Conference 3 or Conference 6 for Meridian 1 proprietary telephones.

LD 10 – Enable Conference 3 or Conference 6 for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA (C6D) C6A	(Deny) allow transfer Class of Service. (Deny) allow six-party conference (C6A requires an XFA Class of Service).

LD 11 – Enable Conference 3 or Conference 6 for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx AO3 AO6	Add a Conference 3 or Conference 6 key (must be key 23 for the M3000). xx = key number.

Feature operation

To add a new party to an established call on a Meridian 1 proprietary telephone, follow these steps:

1 Press **Conference**.
The first party is on hold and you receive a dial tone.

2 Dial the number of the new party.
When the new party answers, you may talk privately.

3 Press **Conference** to include all parties in the call.

4 To add more parties to the conference (up to six, including yourself), repeat steps 1-3.

Note: If you make a mistake while dialing or receive a busy signal, press **Rls** to disconnect. To return to the call, press the key beside the fast flashing indicator.

To add a new party to an established call on an analog (500/2500 type) telephone, follow these steps:

1 Flash the switchhook.
You hear three beeps followed by dial tone. The first party is on hold.

2 Dial the telephone number of the person to be included in your call.
When the call is answered, you may talk privately with the new party.

3 Flash the switchhook to include all parties in the call.

4 To add more parties to the conference (up to six, including yourself), repeat steps 1-3.

Note: If you make a mistake while dialing or receive a busy signal, flash the switchhook to return to the original caller.

Console Operations

Contents

This section contains information on the following topics:

Feature description 1169
Operating parameters 1170
Feature interactions 1171
Feature packaging 1171
Feature implementation 1171
Feature operation 1173

Feature description

Console Operations consists of Console Presentation and Queue Thermometer.

Console Presentation

This part of the feature makes it possible to present a call to a certain Incoming Call Indicator (ICI) key only to specified consoles of a customer or an Attendant Console Group (ACG).

Those consoles are configured (LD 15 option PSA/PSD) to have presentation status for the ICI key to handle the following call treatments:

- If a call is not automatically presented to an idle console, it is indicated on the appropriate ICI key on all consoles within the customer or ACG (LD 15 option MTI/CUI).

- When all attendants with presentation status for a certain ICI key within a group, customer or ACG are in position busy, then a call to that ICI key will be presented to any other console within the group or only presented to the last console that is not in Night Service in a group (LD 15 option RECA/RECO).
- When a ACG is in Night Service, a call to that ACG is redirected to the customer or trunk night DN, or to a night ACG (LD 15 option NCA/ NCD).

Queue Thermometer

This part of the feature applies to a special console which has four single-digit numeric displays. Each display can be configured to show the number of attendant calls in queue for specified ICI keys of a customer or an ACG (LD 15 option MTI/CUI).

A queue thermometer display can also be configured to show the number of calls to those ICI keys that are not individually displayed on any other display of that console.

Optionally, this sum may exclude calls to ICI keys to which inter-attendant calls, recalls and metered calls are presented (LD 15 option DRT/DRE).

Operating parameters

Console Presentation

When Multi-tenant Service (MTS) is in use, the following limitations apply when using Console Operations:

- All attendants must belong to an ACG
- All attendants must belong to only one ACG, and
- No attendants must belong to ACG 0.

Feature interactions

Console Presentation

Departmental Listing Directory Number

Departmental Listing Directory Number is a way of directing attendant calls. The feature has some similarities to MTS, but it overrides Multi-tenant Service (MTS) and is therefore not affected by Console Presentation.

Listed Directory Numbers, Network Wide

Console Operation makes it possible for each console to select which ICI call types will be presented to the console. Network wide LDN does not work with the Console Presentation feature because it is not supported by NAS. Console Operation can, however, be configured with two additional LDNs.

The queue thermometer indicates how many calls are in the queue for a certain ICI key. An ICI key can correspond to more than one ICI type. Even though the ICI type of a call may be different with or without this feature active, it will not interact with queue thermometer operations.

Feature packaging

Console Operations (COOP) package 169.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the Console Presentation feature.

2 LD 15 – Configure the Queue Thermometer feature.

Console Presentation

LD 15 – Configure the Console Presentation feature.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	ATT	Attendant console options
...		
- OPT	(PSD) PSA	Presentation Status selection allowed (denied) on Attendant Consoles.
	(CUI) MTI	ICI lamps show Multi-tenant Service (MTS) Attendant Console Group (AGP) information for incoming calls.
	(RECO) RECA	Attendant calls will be redirected when there is no presentation status to other consoles in the console group; RECO when all consoles are busy, RECA when all but one console is busy.
	(NCD) NCA	When an Attendant Console Group is in Night Service, redirection of attendant calls is allowed (denied).

Queue Thermometer

LD 15 – Configure the Queue Thermometer feature.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	CDB	Customer Data Block.
...		
- OPT	(DRE) DRT	Queue thermometer includes (excludes) Inter-Attendant calls, Recalls and Metered calls.

Feature operation

No specific operating procedures are required to use this feature.

Console Presentation Group Level Services

Contents

This section contains information on the following topics:

Feature description 1175
Operating parameters 1176
Feature interactions 1176
Feature packaging 1177
Feature implementation 1178
Feature operation 1180

Feature description

A Console Presentation Group (CPG) is a subset of the consoles configured for a customer. A CPG handles attendant calls from one or more tenants and incoming trunk calls on one or more routes. CPG improves functions for the following CPG Level Services:

- Attendant Overflow Positions (AOP)
  AOP DN and waiting time threshold can be specified for each CPG.
- Call Waiting Indication
  Count thresholds, timers, and buzz options can be defined for each CPG.
- Incoming Call Identification (ICI)
  ICI keys can be defined for each CPG. Attendants see only those ICI definitions for their own CPG.

- Listed Directory Numbers (LDN)
  Each CPG allows four LDNs.
- Night Service (NSVC)
  Each CPG can go into Night Service mode independent of the other groups.
- Recorded Announcement (RAN)
  Each CPG can have its own recorded overflow announcements.

Operating parameters

Console Presentation Group (CPG) services and Departmental Listed Directory Numbers (DLDN) are mutually exclusive at the customer level. That is, DLDNs can be equipped on the same system with Console Presentation Groups (CPGs), but not enabled for the same customer group at the same time.

Feature interactions

Attendant Administration

Attendants can dial the access code and activate the Administration mode. In this mode, they can modify the configuration of any telephone for this customer.

Attendant Secrecy

The Secrecy option specified for a customer applies to all attendants for that customer.

Attendant Supervisory Console

The supervisory console specified for a customer belongs to one CPG. In the Supervisory mode, ICI indicators show only the information for ICIs in that CPG. Thresholds specified in the Customer Data Block apply only to the CPG where that console resides, and do not effect any other CPG.

Call Park

Parked calls recall to the attendant who parked them. If that Attendant Console goes into Position Busy mode, the call recalls to an attendant in the same CPG as the original.

If the attendant goes into Night Service while a call is parked, the recall is presented to the Night DN defined for that CPG. If an attendant goes into Night Service while the recall is in the attendant queue, it stays in the attendant queue until the call is abandoned.

Tenant access checking between the set (A) who picks up a parked call and the party (B) who parked the call, is enforced as follows:

- If B is a set, tenant-to-tenant access must be allowed between A and B.
- If B is an attendant, A and B must belong to the same CPG for tenant-to-tenant access.
- If access is denied, set A (who intends to pick up the access-denied parked call) receives a blocking tone.

Network-Wide Listed Directory Number

CPG does not work with the network part of Network-Wide Listed Directory Number (LDN) because CPG is not supported by Network Attendant Service, which the network part requires. This feature does, however, provide two additional LDNs for each Console Presentation Group.

Night Key for Direct Inward Dialing Digit Manipulation

The Day/Night table can be activated with the DRC key by any attendant in the Console Presentation group.

Feature packaging

Console Presentation Groups (CPGS) package 172 requires:

- Multi-Tenant Service (TENS) package 86

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 93 – Enable Console Presentation Group (CPG).

2 LD 93 – Assign Attendant Consoles to a presentation group.

3 LD 93 – Assign tenants to an attendant group number.

4 LD 93 – Assign a route to an attendant group number.

5 LD 93 – Add Console Presentation Group features.

LD 93 – Enable Console Presentation Group (CPG).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	TENS	Multi-Tenant data block.
CUST	xx	Customer number, as defined in LD 15
CPGS	YES	Enable CPG Level Services.

LD 93 – Assign Attendant Consoles to a presentation group.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CPG	Console Presentation Group data block.
CUST	xx	Customer number, as defined in LD 15
AGNO	0-63	Attendant Console group number.
ANUM	1-63 1-63	Attendant Console numbers.

LD 93 – Assign tenants to an attendant group number.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	TCPG	Tenant to Console Presentation Group data block.
CUST	xx	Customer number, as defined in LD 15
TEN	1-511	Tenant number.
AGNO	0-63	Attendant Console group number.

LD 93 – Assign a route to an attendant group number.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RCPG	Route to Console Presentation Group data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
AGNO	0-63	Attendant Console group number.

LD 93 – Add Console Presentation Group features.

Prompt	Response	Description
REQ	NEW CHG	Add, or change Multi-Tenant Service for a customer.
TYPE	CPGP	Console Presentation Group level parameters.
CUST	xx	Customer number, as defined in LD 15
CPG	1-63	Console Presentation group number.
LDN0	xxxx	Listed DN 0.

NIT1	xxxx	First Night Service by Time of Day (NTOD) DN.
TIM1	hhmm	Hour minute for First NTOD DN.
NIT2	xxxx	Second NTOD DN.
TIM2	hhmm	Time for Second NTOD.
NIT3	xxxx	Third NTOD DN.
TIM3	hhmm	Time for Third NTOD DN.
NIT4	xxxx	Fourth NTOD DN.
TIM4	hhmm	Time for Fourth NTOD.
ICI	xx aaa	Incoming Call Indicators (ICI).
AQTT	0-(30)-255	Attendant queuing threshold.
AODN	xxxx	Attendant overflow DN.
CWCL	(0)-255 (0)-255	Number of waiting calls, lower threshold and upper bound.
CWTM	(0)-511 (0)-511	Time for waiting calls, lower threshold and upper bound.
CWBZ	(NO) YES (NO) YES	Call Waiting Buzz. The first field provides a two-second buzz when the upper CWCL or CWTM threshold is exceeded. The second field provides a buzz when the first call enters the queue.

Feature operation

No specific operating procedures are required to use this feature.

Controlled Class of Service

Contents

This section contains information on the following topics:

Feature description 1181
Operating parameters 1181
Feature interactions 1182
Feature packaging 1184
Feature implementation 1184
Feature operation 1186

Feature description

Controlled Class of Service (CCOS) alters the Class of Service restriction levels on telephones that have been defined as CCOS controlling telephones. This applies to Meridian 1 proprietary telephone users designated as CCOS controllers. While CCOS is active, Public Exchange/Central Office or toll calls made from these telephones cannot be completed without first being routed through an attendant.

Meridian 1 proprietary telephones designated as CCOS controlling telephones are assigned a CCOS key/lamp that is used to activate or cancel the system-defined CCOS restriction level on individual DNs.

Operating parameters

Controlling telephones can be any Meridian 1 proprietary telephone.

CCOS controlling telephones must refer to the Prime Directory Number (PDN) when activating or canceling CCOS on other telephones.

Automatic Call Distribution (ACD) agents cannot be restricted by CCOS.

Feature interactions

Authorization Code

The Authorization Code overrides a telephone's CCOS restriction level.

Conference

If CCOS is activated at a telephone involved in a conference call, established Central Office or toll calls are not affected. The CCOS restriction level is applied immediately, and no new calls can be initiated from the conference. The telephone remains in the CCOS active state after the conference is terminated.

Flexible Feature Codes

If Electronic Lock (ELK) is activated, the CCRS Class of Service is used whether Controlled Class of Service (CCOS) is active or not. ELK takes precedence over CCOS. If ELK is deactivated, the set is treated as per existing operation.

When FFC ELKA and a password is entered, this set will use the CCRS Class of Service configured in LD 15. The CCRS Class of Service will always be used whether or not CCOS is currently controlling the set's Class of Service. When FFC ELKD and a password is entered, the set will use the appropriate Class of Service associated with this set. If CCOS is enabled for the set, the associated customer Class of Service is used (that is, CCRS, ECC1, or ECC2). If CCOS is not enabled for this set, the set's own Class of Service is used.

When FFC ELK is deactivated, the set reverts back to the Class of Service as it should be without FFC ELK, instead of always reverting back to the set's Class of Service (that is, if CCOS is enabled, it will use the customer's Class of Service; if CCOS is not enabled, it will use the set's Class of Service).

Hot Line

When a Hot Line DN is on a telephone that has Controlled Class of Service activated, Hot Line calls ignore the imposed Class of Service if the System Speed Call (SSC) package is present and the Hot Line list is given an adequate Network Class of Service (NCOS) for the override

Multiple Appearance Directory Number

CCOS restriction levels are activated or canceled on controlled telephones through their Prime Directory Number (PDN). When the PDN of a Meridian 1 proprietary telephone is made CCOS active, all DNs on that telephone are also restricted. If the DN is a PDN on other telephones, those telephones are also restricted (if they have CCSA Class of Service).

Room Status

You can change the access restrictions for room telephones from the BGD or from a telephone equipped with a Room Status key (RMK).

Scheduled Access Restrictions

During normal hours, CCOS restrictions override normal telephone restrictions. During off-hour periods or times when a Scheduled Access Restrictions (SAR) LOCK is in effect, however, Scheduled Access Restrictions apply. When the LOCK or off-hour period ends, CCOS restrictions continue to apply until they are removed or SAR becomes effective again. Whether a CCOS controller or electronic lock is used to activate CCOS, there is no indication to the user when Scheduled Access Restrictions are in effect, overriding CCOS restrictions. A telephone defined in LD 10 or 11 or a trunk defined in LD 14, which is assigned an SAR group number, has its Class of Service defined by the SAR schedule of its SAR group.

Station Category Indication

The Controlled Class of Service (CCOS) feature has priority over SCI. A station's SCI category is suppressed when CCOS is active, and calls to the attendant DN carry the CCOS class defined in the database.

Feature packaging

The (CCOS) package must be equipped to activate the Controlled Class of Service feature.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable CCOS for a customer.

2 LD 11 – Allow CCOS on Meridian 1 proprietary telephones.

3 LD 10 – Allow CCOS on analog (500/2500 type) telephones.

4 LD 11 – Change CCOS controlling telephone assignments on Meridian 1 proprietary telephones.

LD 15 – Enable CCOS for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CCS	Controlled Class of Service options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CCRS	UNR CUN CTD TLD SRE FRE FR1 FR2	Unrestricted. Conditionally unrestricted. Conditionally toll-denied. Toll-denied. Semi-restricted. Fully restricted. Fully restricted 1. Fully restricted 2.

LD 11 – Allow CCOS on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CCSD), CCSA	(Deny) allow CCOS.

LD 10 – Allow CCOS on analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CCSD) CCSA	(Deny) allow CCOS.

LD 11 – Change CCOS controlling telephone assignments on Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx COS	Assign CCOS controlling key.

Feature operation

To activate CCOS, follow these steps:

1 Press **CCOS**.

Note that this is a toggle: If CCOS is already active, pressing the key will change the CCOS state to inactive. Check the CCOS lamp to determine if CCOS is already active.

2 Dial the Prime Directory Number (PDN) of the telephone to be changed and press CCOS.

3 Press **Rls**.

To deactivate CCOS, follow these steps:

1 Press **CCOS**.

2 Dial the PDN of the telephone to be returned to its original Class of Service and press CCOS.

3 Press **Rls**.

Controlled Class of Service, Enhanced

Contents

This section contains information on the following topics:

Feature description 1187
Operating parameters 1187
Feature interactions 1188
Feature packaging 1189
Feature implementation 1189
Feature operation 1192

Feature description

Enhanced Controlled Class of Service (ECCS) allows a controller or Attendant Console to alter the Class of Service (CLS) restriction levels of other Controlled Class of Service (CCOS) telephones. The feature allows two customer-defined levels of restriction. In addition, the CCOS key can be assigned to an Attendant Console and M3000 telephones as a programmable key.

Operating parameters

Controlling telephones can be any Meridian 1 proprietary telephone.

A CCOS controlling telephone must refer to the Prime DN when activating or canceling CCOS on other telephones.

Automatic Call Distribution (ACD) agents cannot be restricted by CCOS.

On M3000 telephones, the CCOS key can be assigned as a programmable key (0-5 only).

This feature is applicable only when the CLS lamp is lit on the controlling telephone.

The CLS key on an Attendant Console can be used only on an idle loop. (The loop lamp is lit; source and destination lamps are dark.)

Feature interactions

Attendant Administration

This feature cannot change Controlled Class Service restrictions (CCRS), ECC1 or ECC2, but can assign CLS keys to certain telephones.

Attendant Supervisory Console

When the attendant is in the supervisory mode, CCOS programming is prohibited.

Authorization Codes

The Authorization Code can override a telephone's CCOS restriction level.

Conference

If CCOS is activated at a telephone on a conference call, established Public Exchange/Central Office or toll calls are not affected. The CCOS restriction level is applied immediately; however, no new calls can be initiated from the conference. That telephone remains in the CCOS state after the end of the conference.

Coordinated Dialing Plan

The internal DN is used for programming the CLS level for Coordinated Dialing Plan (CDP) from the controlling telephone.

Multiple Appearance Directory Number

All CCOS restriction levels are activated and canceled from the Prime Directory Number (PDN) for CCOS controlling telephones. The PDN for a Meridian 1 proprietary telephone is made CCOS active, and all DNs for that telephone are restricted as well. If that DN is a PDN on other telephones, they are also restricted (if they have CCSA Class of Service).

Pretranslation

The DN used to program the CCOS should be the actual DN before pretranslation. When programming CCOS, the DN entered is not pretranslated.

Feature packaging

Enhanced Controlled Class of Service (ECCS) package 173 requires:

- Controlled Class of Service (CCOS) package 81.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Define the Class of Service restrictions for the system.

2 LD 11 – Assign keys for the controller telephone.

3 LD 10 – Configure controlled analog (500/2500 type) telephones.

4 LD 11 – Configure the controlled Meridian 1 proprietary telephones.

5 LD 12 – Assign ECCS keys for Attendant Console.

LD 15 – Define the Class of Service restrictions for the system.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CCS	Controlled Class of Service options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CCRS		CCOS restrictions.
	(UNR) CTD CUN FRE FR1 FR2 SRE TLD	Unrestricted service. Conditionally Toll Denied. Conditionally Unrestricted. Fully Restricted. Fully Restricted level 1. Fully Restricted level 2. Semi-Restricted. Toll Denied.
- ECC1	xxx	Enhanced Controlled Class of Service, Level 1. xxx = (UNR), CTD, CUN, FRE, FR1, FR2, SRE, TLD.
- ECC2	xxx	Enhanced Controlled Class of Service, Level 2. xxx = (UNR), CTD, CUN, FRE, FR1, FR2, SRE, TLD.

Note: Input restrictions apply when CCSA is active. When CCSA is inactive, the telephone has the CLS assigned in LD 10/11.

LD 11 – Assign keys for the controller telephone.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx COS	Key number for CCOS key on controller telephone (for the M3000, the key must be 0-5).

LD 10 – Configure controlled analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CCSD) CCSA	(Deny) allow CCOS.

LD 11 – Configure the controlled Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CCSD) CCSA	(Deny) allow CCOS.

LD 12 – Assign ECCS keys for Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx COS	Key number for CCOS controller key on Attendant Console. xx = key number (must be greater than 1).

Feature operation

To activate Enhanced Controlled Class of Service (ECCS) from a Meridian 1 proprietary telephone with the feature currently inactive, follow these steps:

1 Press **CCOS** to begin the activation sequence.

Note that this is a toggle: if CCOS is already active, pressing the key will change the CCOS state to inactive. Check the CCOS lamp to determine if CCOS is already active.

2 Dial the PDN of the telephone to be changed and press **CCOS**. The controlling telephone's display, if equipped, shows the DN of the changed telephone and a 0 (zero).

3 To select ECC1, dial # 1.
Note that the octothorpe (#) is required. The controlling telephone's display, if equipped, shows the DN of the changed telephone and a 1.

To select ECC2, dial # 2.
Note that the octothorpe (#) is required. The controlling telephone's display, if equipped, shows the DN of the changed telephone and a 2.

4 Press **Rls**.

To activate ECCS from an Attendant Console, follow these steps:

1 Select an idle loop key.

2 Press **CCOS**.

Note that this is a toggle: If CCOS is already active, pressing the key will change the CCOS state to inactive. Check the CCOS lamp to determine if CCOS is already active.

3 Dial the PDN of the telephone to be changed and press **CCOS**. The console's display shows the DN of the changed telephone. A 0 (zero) is displayed if the telephone is active in the original CCOS mode.

If the telephone does not have CCOS or ECCS active, the console does not acknowledge that you have successfully entered a valid CCOS DN.

4 To select ECC1, dial # 1.
Note that the octothorpe (#) is required. The console's display shows the DN of the changed telephone and a 1.

To select ECC2, dial # 2.
Note that the octothorpe (#) is required. The console's display shows the DN of the changed telephone and a 2.

5 Press **Rls**.

To deactivate Enhanced Controlled Class of Service (ECCS), follow these steps:

1 Select an idle loop key.

2 Press **CCOS**.

3 Dial the PDN of the telephone to be returned to its original Class of Service and press **CCOS**.

4 Press **Rls**.

Meridian 1, Succession 1000,
Succession 1000M

Features and Services

Book 1 of 3 (A to C)

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Departmental Listed Directory Number

Contents

This section contains information on the following topics:

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Feature interactions 75
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Feature implementation 77
Feature operation 79

Feature description

The Departmental Listed Directory Number (DLDN) feature allows specified telephones sharing the same numbering plan to belong to one subgroup out of a possible six subgroups within a system customer group. Each Departmental Listed Directory Number (DLDN) subgroup is identified by one of the customer's Listed Directory Numbers (LDNs). Calls to specific Listed Directory Numbers (LDN), or dial-0 calls from subgroup telephones, are directed to the Attendant Console or consoles assigned to that LDN.

When the Departmental Listed Directory Number (DLDN) feature is implemented, a departmental Attendant Console is presented with calls from the following sources:

- Incoming external trunk calls routed to the LDN from:
  - an auto-terminate trunk (CO, FX, or WATS) whose Auto-Terminate Number (ATDN) is the LDN

— a Direct Inward Dialing (DID) trunk whose DID number is the same as the LDN

- Calls that originate from internal telephones or TIE trunks when:

  — a telephone user dials the LDN

  — a telephone user associated with a departmental Attendant Console dials 0, or

  — a TIE-line user dials the LDN.

The DLDN feature associates Attendant Consoles with an LDN. Up to 63 Attendant Consoles can be associated with one LDN.

For call distribution purposes, all Attendant Consoles within a subgroup are made members of a circular list. When a call is received, it is presented to the next listed console after the one that was last offered a call, thus ensuring that LDN calls are distributed in an equitable way. LDN calls, dial-0 calls, and associated timed recalls are serviced according to a circular list for the particular LDN.

On receiving an LDN type call, the system searches for an idle Attendant Console and tests whether or not that console is configured to answer a call for the dialed Directory Number (DN). If the Attendant Console is not configured to answer calls for that LDN, the next idle Attendant Console is tested. If an Attendant Console that can answer the call is found, the call is presented with the appropriate Loop and Incoming Call Indicator (ICI) lamps lit. If no idle Attendant Console for the LDN is found, the call is placed in the incoming call queue for all Attendant Consoles within the customer group.

The Call Waiting indication is provided to all Attendant Consoles within the customer group. If an Incoming Call Indicator (ICI) key has been provisioned for the LDN, a lamp indication (with no buzz) is provided to all idle Attendant Consoles within the customer group and may be answered by pressing the appropriate key.

When an attendant presses the Release key, the system checks to see if there are any calls waiting in the queue. If there are calls waiting, it tests whether or not the Attendant Console, if it is next in the circular list, can answer the first call in the queue. If the call can be answered, it is presented to the Attendant Console. Otherwise it is put back into the queue and another call is sought. If no calls for the LDN are found, the Attendant Console is idled and the Release lamp is lit.

Operating parameters

An optional assignment of ICI keys is allowed to provide a visual indication of the LDN (LD 15).

If the DN Expansion package is equipped, all LDNs can have up to seven digits.

Feature interactions

Attendant Overflow Position

LDN calls that have been waiting in the queue longer than the specified threshold period will be routed to the Attendant Overflow Position.

Attendant Position Busy

If all Attendant Consoles in a LDN group are in position busy, calls to that LDN are not automatically presented to any Attendant Console in the customer group and will enter the attendant queue for that customer group. Other attendants outside the LDN queue may only answer LDN calls in the attendant queue by pressing the relevant LDN ICI key, if configured. No buzz is provided as the call is in the attendant queue and not the loop key.

Attendant Supervisory Console

The supervisory capabilities extend to all Attendant Consoles defined within the customer group. The Attendant Console serving as supervisor should be a member of every DLDN group so that it can serve all groups when operating in the Normal mode.

Call Forward Busy
Call Forward No Answer
Call Forward

Call Forward No Answer to the attendant and Call Forward Busy operate like Call Forward to 0, and are routed to any idle Attendant Console in the customer group.

Centralized Attendant Service

LDN calls are not screened for Centralized Attendant Service (CAS). When a CAS key is pressed at a CAS remote Attendant Console, LDN calls will be handled at the CAS main as if the DLDN feature did not exist.

Console Operations

DLDN is a way of directing attendant calls. The feature has some similarities to MTS, but it overrides Multi-tenant Service (MTS) and is therefore not affected by Console Presentation.

Directory Number

With the Network-Wide LDN feature, telephones using DLDN have access to two additional LDNs, even though DLDN is not supported over a network.

Interdepartmental Attendant Transfers

Interdepartmental Attendant Transfers operate normally, except that if there is a recall, it will be to the appropriate department rather than to the last attendant that extended the call.

Listed Directory Numbers, Network Wide

Departmental LDN is not supported over the network; however, this feature does provide two more LDNs for the DLDN feature.

Multiple Console Operation

Departmental Listed Directory Number (DLDN) supports the assignment of 63 consoles per DLDN.

Network-Wide Listed Directory Number

DLDN is not supported over a network; however, Network-Wide LDN provides two additional LDNs for DLDN.

Night Service

DLDN does not affect Night Service (including TAFAS). Calls presented to the LDN from an external source will queue for the night bell. All other attendant calls receive busy treatment if the night Directory Number (DN) is busy.

Feature packaging

Departmental Listed Directory (DLDN) package 76 has no other package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Enable the Departmental Listed Directory Number feature for a customer.
2. LD 10 – Configure Departmental Listed Directory Number for analog (500/2500 type) telephones.
3. LD 11 – Configure Departmental Listed Directory Number for Meridian 1 proprietary telephones.

LD 15 – Enable the Departmental Listed Directory Number feature for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	LDN	Departmental Listed Directory Numbers
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	NLDN	Network wide LDN allowed. XLDN = Network wide LDN denied (default).
- DLDN	(NO) YES	(Disable) enable DLDN.
- LDN0	xxxx	Listed Directory Number Zero.
- LDA0	1 - 63 ALL	Attendant Consoles associated with LDN 0.
- LDN1	xxxx	Listed Directory Number One.

- LDA1	1 - 63 ALL	Attendant Console number associated with LDN 1.
- LDN2	xxxx	Listed Directory Number Two.
- LDA2	1 - 63 ALL	Attendant Console number associated with LDN 2.
- LDN3	xxxx	Listed Directory Number Three.
- LDA3	1 - 63 ALL	Attendant Console number associated with LDN 3.
- LDN4	xxxx	Listed Directory Number Four.
- LDA4	1 - 63 ALL	Attendant Console number associated with LDN 4.
- LDN5	xxxx	Listed Directory Number Five.
- LDA5	1 - 63 ALL	Attendant Console number associated with LDN 5.
- ICI	xx LD0 xx LD1 xx LD2 xx LD3 xx LD4 xx LD5	Incoming Call Indication for Listed Directory Numbers Zero to Five (xx = key number 00-19).

Note: To remove an LDN, enter an X before the Directory Number. An LDN cannot be removed if any Attendant Consoles are associated with it. To remove an associated Attendant Console, enter an X at the LDA prompt before the attendant number.

LD 10 – Configure Departmental Listed Directory Number for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
LDN	(NO) 0-3	Telephone associated with LDN (0-3 or none). Choose NO to remove this telephone from the group.

LD 11 – Configure Departmental Listed Directory Number for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
LDN	(NO) 0-3	Telephone associated with LDN (0-3 or none). Choose NO to remove this telephone from the group.

Feature operation

No specific operating procedures are required to use this feature.

Dial Access to Group Calls

Contents

This section contains information on the following topics:

Feature description 81
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Feature interactions 82
Feature packaging 83
Feature implementation 83
Feature operation 84

Feature description

This feature allows attendants and users of analog (500/2500 type) telephones, and Meridian 1 proprietary telephones to make a Group Call by dialing a Flexible Feature Code (FFC). telephone users may continue to use a Group Call key. The customer can define whether or not the originator of the Group Call has control of the active call. In the Group Call List, if GRPC = YES, the originator has control: when the originator goes on hook, the call is terminated. If GRPC = NO and the originator goes on hook, the Group Call acts like a conference call: the call remains active until all members go on hook.

For more information on group calls, see “Group Call” on page 429 in this book.

Operating parameters

All group stations must have Warning Tone Allowed (WTA) Class of Service.

Because analog (500/2500 type) telephones have no lamp state, there is no indication to the call originator that all group members have answered.

If a Group Call is originated using an FFC from a DN key of a Meridian 1 proprietary telephone, or a loop key on an Attendant Console, the DN lamp state does not display the status of the Group Call.

A Group Call member that has disconnected from the call cannot be reconnected to the call.

Feature interactions

The following features cannot be applied to a Group Call:

- Call Forward No Answer
- Call Forward Busy
- Call Join
- Call Park
- Conference
- Hunting
- Privacy Release, and
- Ring Again.

AC15 Recall: Transfer from Norstar

If Norstar sends a recall signal in order to initiate a consultation, the consultation will not be authorized because it is not possible to put a group call on hold. It is however possible to transfer a party to a group call using an AC15 trunk.

On Hold on Loudspeaker

If a group call is initiated from a set with Dealer Allowed Class of Service, the conference is built up on the assigned loop of the loudspeaker or speech monitor system channel since this is a potential On Hold on Loudspeaker call.

Feature packaging

Dial Access to Group Call requires the following packages:

- Group Call (GRP) package 48
- Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 18 – Configure the Group Call List table for Group Call control.

2 LD 57 – Configure Flexible Feature Codes for Group Calls.

LD 18 – Configure the Group Call List table for Group Call control.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	GRP	Group Call data block.
CUST	xx	Customer number, as defined in LD 15
GRNO	0-63	Number of the Group Call list.
STOR	xx yyy...y	Group member number (xx) and associated DN (yyy...y).

LD 57 – Configure Flexible Feature Codes for Group Calls.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
GRPF	xxxx	Group Call code.
- GRCL	0-63	Group Call List number.

Feature operation

To make a Group Call,

- Press the **Group Call** key. All group members are automatically called. The LCD indicator beside the Group Call key flashes until all members have answered. Then it lights steadily.

or

- Dial the Group Call FFC. All group members are automatically called. When an originating station makes a Group Call using an FFC, all idle stations in the group are rung. Busy stations are given Call Waiting or Camp-On, if equipped, along with a special warning tone.

Dial Intercom

Contents

This section contains information on the following topics:

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Feature operation 92

Feature description

Dial Intercom (DI) allows a customer to arrange stations within the system into separate Dial Intercom Groups (DIGs). A total of 100 stations can belong to each Dial Intercom Group (DIG). One-digit dialing is required for a Dial Intercom Group (DIG) of up to 10 stations, and two-digit dialing is required for a DIG of up to 100 stations.

Meridian 1 proprietary telephones can be equipped with a separate DIG key/lamp pair for each DIG of which it is a member. Single-line telephone users can belong to only one DIG and may not have any non-DIG Directory Numbers (DNs).

Voice or ring may be specified on a DIG basis for Meridian 1 proprietary telephones. If voice is specified, an idle station rings once for two seconds. The calling party is then connected and may make a voice announcement. If ring is implemented, normal ringing is received until the called party answers. This feature provides the option of an announcement or a two-way speech path.

The ring option must be used if a 500 telephone is a member of the group.

Distinctive ringing for Dial Intercom

This feature allows a user to differentiate between an incoming call and a Dial Intercom (DI) call. The Dial Intercom (DI) ringing has a different cadence than the regular Directory Number (DN) ringing and Distinctive Ringing.

Distinctive Ringing for DI is assigned on a per-customer basis. The cadence is 0.5 sec. on and 0.5 sec. off, repeatedly.

Dial Intercom Handsfree Voice Call

Dial Intercom Handsfree Voice Call can be used with the following telephones: M2112, M2317, and M2616.

Handsfree Voice Call provides the option of configuring VCC/DIG (with voice option) to be answered in either handsfree mode or loudspeaker only mode. Calls answered in handsfree (HVA) mode establish a two-way voice path, while those answered in loudspeaker only (HVD) mode establish only a one-way voice path from the calling telephone to the destination telephone.

Dial Intercom Handsfree Voice Call applies only to voice option DIG calls.

Operating parameters

A maximum of 2046 DIGs can be established per customer.

Calls are restricted to stations within the DIG only. Trunks cannot be accessed using the DIG key, and cannot be added to a DIG call using the Conference feature.

A DIG member number must be a single appearance Directory Number (DN) within a specified DIG.

DI analog (500/2500 type) telephones cannot dial the attendant or be dialed by the attendant.

A DI telephone cannot be assigned a member number that conflicts with the Special Prefix (SPRE) code. In the case of double-digit DIG values, the first digit cannot be the same as the SPRE code. For example, if the SPRE code is 7, the member number cannot be 7 or any number from 70 through 79. A two-digit SPRE code, such as 77, allows 99 DIG member numbers (00, 01-76, and 78-99). With no SPRE code defined, 100 DIG members are possible.

Call Transfer and Conference cannot take place to telephones outside the DIG.

Handsfree Voice Call allowed/denied is set at the system level and can only be used with digital telephones that have handsfree capabilities (such as 2112, 2317, and 2616), and requires Class of Service Handsfree Allowed (HFA) on the destination telephone, which is set at the telephone level.

Basic Rate Interface (BRI), M3000, and SL-1 telephones do not support the Handsfree feature.

Feature interactions

Auto Answer Back (AAB)

This feature is not affected by the Handsfree Voice Call feature.

Autodial
Speed Call

The Dial Intercom code may be dialed using Autodial or Speed Call.

Automatic Line Selection

A Dial Intercom DN is selected by Incoming Ringing Line Selection and Outgoing Line Selection.

Call Forward
Call Waiting

The Call Forward and Call Waiting features do not apply to a Dial Intercom appearance.

Call Party Name Display

The display on telephones connected by Dial Intercom shows the group member's DIG number plus Call Party Name Display information.

Call Pickup

Call Pickup may be used by Meridian 1 proprietary telephones if the telephones are all in the same DIG and Call Pickup Group and the ring option is specified for the DIG.

Call Pickup Network Wide

The Dial Intercom feature is not supported network wide. Any pickup attempt from a distant node to a local intercom call will be rejected, because the far-end user is considered as not being in the same intercom group.

Conference
Call Transfer

When using Conference or Transfer, the voice option is not provided if the call is terminated before the conference or transfer is completed. If an analog (500/2500 type) telephone is part of a Dial Intercom Group (DIG), the user of the telephone can conference only with another user whose telephone is within the same Dial Intercom Group (DIG).

Digit Display

The digit display will be cleared when the DIG key is pressed. When the user dials the DI code, the digits of the code are displayed. When the call is answered, the DI code of the calling party appears on the display of the called party.

If either party presses the Release key or goes on hook during a DIG call, the displays of both parties are cleared. If either party presses the Hold key, the display of the holding station is cleared but the display of the other station remains unchanged. When the held call is reestablished, the holding station redisplays the DIG number of the other party.

Display of Calling Party Denied

Display information on sets that are involved in a Dial Intercom Group (DIG) call is based on the individual Class of Service of each set. If a DN is denied for a set involved in a DIG call, the DIG number for that set is replaced by one dash (–) in the case of 10 DIG stations. For 100 DIG stations, the DIG number is replaced by two dashes (– –).

Hot Line

The analog (500/2500 type) Hot Line telephones cannot be members of Dial Intercom Groups (DIGs).

Station features

DI can be used in combination with the following features:

Feature	Meridian 1 proprietary telephones	Analog (500/2500 type) telephones
Autodial	•	
Speed Call	•	
Digit Display	•	
Make Set Busy	•	
Override	•	
Release	•	
Hold	•	
Call Pickup	•	•
Conference	•	•
Call Transfer	•	•
Ring Again	•	•

Tones, Flexible Incoming

For Dial Intercom Group (DIG) calls with the voice (V) option, if the telephone receiving the call is busy, the user hears one buzz followed by a flashing indicator. This is how DIG works with or without FIT.

Feature packaging

Dial Intercom (DI) package 21 has no other package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Dial Intercom for a customer.

2 LD 10 – Configure Dial Intercom for analog (500/2500 type) telephones.

3 LD 11 – Configure Dial Intercom for Meridian 1 proprietary telephones.

4 LD 15 – Configure Handsfree Voice Call for the system.

LD 15 – Enable Dial Intercom for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- DGRP	0-2046	Maximum number of DIGs that can be defined for the customer. The maximum number of DIGs allowed is 2046.

LD 10 – Configure Dial Intercom for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change. Single line telephones cannot have both a Dial Intercom Group number and a standard DN. To add this feature, you must remove the telephone from the database and build it again, as a Dial Intercom Group member.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

DES	a...x	ODAS set designator. a...x = one-to-six character alphanumeric designator.
CUST	xx	Customer number, as defined in LD 15
DIG	xxxx yy	xxxx = Dial Intercom group number (0-2046). yy = member number (0-99) within the group. The maximum number of DIGs allowed is to 2046.

LD 11 – Configure Dial Intercom for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx DIG aaa bb c	Add a Dial Intercom key, where: xx = key number aaa = group number (0-2046) bb = member number (0-99), and c = r (ring) or v (voice). The maximum number of DIGs allowed is to 2046.

LD 15 – Configure Handsfree Voice Call for the system.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(HVD) HVA	Handsfree Voice Call (Denied) Allowed.

Feature operation

An example of a Dial Intercom call is listed below.

Dial Intercom Call

To make a Dial Intercom call:

1 Lift the handset and dial the **Intercom** key.

2 Dial the one- or two-digit code for the DIG member.

If your phone and the phone you are calling are configured for the voice option, you can deliver a voice message after two seconds of ringing.

To answer a Dial Intercom call when you are on a line other than your DIG line:

1 Release the current call or place it on hold.

2 Press **Intercom**.

Dial Intercom Handsfree Voice Call

Examples of both Handsfree Voice Call options are listed below.

HVA option

The originating telephone (telephone A) places a DIG call to the destination telephone (telephone B).

- Telephone B rings once.
- After one ring, telephone B automatically answers the call in Handsfree mode.

  The DN and handsfree LCDs are lit and a two-way voice path is established.

HVD option

Telephone A places a call to telephone B.

- Telephone B rings once.
- After one ring, telephone B automatically answers the call in loudspeaker only mode.

  The DN LCD is lit and the handsfree LCD remains dark, establishing a one-way voice path from telephone A to telephone B. At this point, telephone A is unable to hear the person at telephone B.

  To establish a two-way voice path, telephone B must either go off hook, or press the Handsfree button.

Dial Pulse/Dual-tone Multifrequency Conversion

Contents

This section contains information on the following topics:

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Operating parameters 96
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Feature packaging 96
Feature implementation 96
Feature operation 96

Feature description

With the Dial Pulse/Dual Tone Multifrequency Conversion feature, Dial Pulse (DP) signals from analog (500/2500 type) telephones, Dial Pulse (DP) TIE lines, Meridian 1 proprietary telephones, or Attendant Consoles are automatically converted to Dual-tone Multifrequency (DTMF) signals for transmission over trunks equipped for Dual-tone Multifrequency (DTMF) service. Dual-tone Multifrequency (DTMF) signals from single-line 2500 telephones are automatically converted for transmission over rotary-dial-only trunks, such as TIE lines. This eliminates the need for duplicate dials.

DTMF calling allows the use of 2500 telephones, equipped with push-button dials, to transmit digits through audible tones to the system equipment. This feature provides the ability to use any combination of telephones. However, 2500 telephones cannot use DTMF to control dictation equipment when the dictation trunk is specified as Dial Pulse (DP).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Dial Tone Detection

Contents

This section contains information on the following topics:

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Feature description

The Dial Tone Detection (DTD) feature is needed because the first digit cannot be sent until the dial tone is detected on calls to a Public Switched Telephone Network (PSTN). This avoids the outpulsing of digits before the PSTN is ready to accept them, thus avoiding either the loss of digits or the misrouting of calls. The possibility of circumventing code dialing restrictions is also minimized by the feature.

The feature is configurable on a route basis for all types of routes.

The time-out for the route is statistically averaged over the last eight times that Dial Tone Detection was involved. Either the running-average time or the pre-overlay programmed minimum time is used as the trunk time out, whichever is greater. Dial Tone Detection can be invoked every time an outgoing trunk route is selected, regardless of the selected feature.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Dial tone detection is supported in the CIS, but with the limitation of low reliability of the tone provided by the Public Exchange.

ISDN Semi Permanent Connections for Australia

To convey D-channel signaling over an ISPC link, the route associated to the link at the system configured as MASTER must detect a dialtone.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Dial Tone detectors are supported with the limitations of the reliability of the tone provided by the Public Exchange.

Feature packaging

Dial Tone Detector (DTD) package 138.

Feature implementation

Gather data for each customer's number to be configured for the DTD feature.

Task summary list

The following is a summary of the tasks in this section:

1 LD 13 – Create or modify data blocks for Digitone Receivers.

2 LD 16 – Create or modify data for trunk routes.

3 LD 17 – Modify the system hardware and software parameters.

LD 13 – Create or modify data blocks for Digitone Receivers.

Prompt	Response	Description
...		
TYPE	DTD	Dial Tone Detection.

LD 16 – Create or modify data for trunk routes.

Prompt	Response	Description
...		
DTD	(NO) YES	Dial Tone Detection is (is not) to be performed on this route.

LD 17 – Modify the system hardware and software parameters.

Prompt	Response	Description
...		
DTDT	NO	No Dial Tone Detection tests are required.

Feature operation

No specific operating procedures are required to use this feature.

Dialed Number Identification Service

The Automatic Call Distribution (ACD) Dialed Number Identification Service (DNIS) shows the last three or four digits of the dialed DNs received from auto-terminated Direct Inward Dialing (DID) and TIE trunks on the display for ACD agents. The maximum number of characters allowed is 27, including spaces.

In telemarketing environments, DNIS can reduce the time needed to serve a call. For example, the dialing plan can be configured so the DNIS digits represent product lines or services. The ACD agent can then answer incoming calls with the correct response

For further information on Dialed Number Identification Service, please refer to *Automatic Call Distribution: Description* (553-3001-351).

Digit Display

Contents

This section contains information on the following topics:

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Feature description

There are two types of Digit Displays: Attendant Console Digit Display and Meridian 1 proprietary telephone Digit Display.

Attendant Console Digit Display

QCW Attendant Consoles can be equipped with either an 8- or a 16-digit display. This display indicates the following:

- Dialed digits
  On attendant-originated calls, Busy Verify (BVR), or Barge-In, the digits dialed by the attendant are displayed. If the dialed number hunts, the Hunt destination and the dialed Directory Number (DN) are displayed. If the dialed number is call forwarded, the forwarded DN and the dialed DN are displayed.

- Incoming calls
  On incoming calls and forwarded Direct Inward Dialing (DID) calls, the trunk access code and member number are displayed. For all station dial-0 calls, the calling station DN is displayed. For recalls, the destination DN is displayed.
- Display Source/Display Destination keys
  Two keys are provided to allow the attendant to display the source and destination numbers for any connection completed through the console.
- Night assignment
  During the assignment of night numbers, the Display Source key may be pressed after the trunk access code and member numbers have been dialed to display the correct night assignment.
- Autodial
  The DN stored against an Autodial key may be displayed by pressing the Autodial key, then the Display Source key. If using an eight-digit display assignment and if the stored DN consists of more than eight digits, the Display Source key must be pressed a second time to display the remainder of the DN. When the Autodial number is changed, the new number may be displayed by pressing the Display Source key.
- Speed Call
  The DN stored against a Speed Call code may be displayed by pressing the Speed Call key, dialing the Speed Call code, and then pressing the Display Source key. When the Speed Call list is changed, an entry may be displayed by pressing the Display Source key.
- Time and Date
  The time may be displayed by pressing the Display Time key on the Attendant Console. The date is displayed by pressing the Display Date key.

Meridian 1 proprietary telephone Digit Display

This feature allows the automatic display of information relevant to normal call processing and feature activation on any Meridian 1 proprietary telephone equipped with a 16-digit display. A key/lamp pair is also provided to enable the station user to obtain information manually, independent of call processing activity.

Time and Date are displayed with an additional Time and Date (TAD) key.

CAUTION
This option should not be used when a Prime DN appears on another telephone as a Prime DN. Severe real-time penalties will occur (ERR040 message).

The following display options are available:

- No Digit Display (NDD)
  This is the default option.
- Automatic Digit Display (ADD)
  This option allows the display of digit information during call processing. ADD allows the automatic display of a calling party number on an incoming call to the Prime DN on a telephone.
- Standard Delayed Display (DDS)
  Provides calling party information, displayed after answer only.
- Touchphone Digit Display (TDD)
  With this option, when a call is presented to a busy M3000 Touchphone, the user of the Touchphone can press the Display key to see the Calling Line Identification information of the new incoming call.
- Tandem Digit Display (TDD)
  With this option, when an incoming call is presented to a busy Meridian 1 proprietary telephone with display, the Calling Line Identification and Call Party Name Display information is automatically displayed on the busy telephone.

Automatic displays will show the following:

- Number dialed
- Number of calling party
- Call Pickup
- Call Waiting party, and
- Time and date.

Press the Display (DSP) key, then the feature key to display information associated with these features:

- ACD in-calls
  If the Display Key is used to view information defined on the ACD DN key of an agent serving multiple queues, then the ACD DN displayed will be the current queue being served if the agent is active on a call. The last queue is served if the agent is not serving an ACD call or the Primary ACD DN if the agent is logged out.
- Autodial number
  When the telephone is inactive and the DSP key is pressed, followed by the autodial key, the number stored against the key will be displayed.
- Autoline
  To display the DN programmed for the Autoline key, the attendant presses the Autoline key when the console is idle or in Position Busy. On an analog console, to display a DN that is longer than eight digits, the attendant presses the display key after pressing the Autoline key.
- Buzz DN
  When the telephone is inactive and the DSP key is pressed, followed by the Buzz DN key, the number stored against the key will be displayed.
- Call Forward party
  When the telephone is inactive and the DSP key is pressed, followed by the Call Forward key, the number stored against the key will be displayed.
- Call Park
  The Park DN of the most recently parked call can be re-displayed on proprietary telephones equipped with displays, a Park key, and a Display key. This is done by pressing the Display key, then the Park key. The attendant can display the last call parked by pressing the Park key when no loop key is active.
- Call Pickup
  To display Call Pickup, press the Display key, followed by the Call Pickup key.

- Call Waiting party
  Pressing the Call Waiting key to answer a waiting call makes that call active. The call can be placed on hold by pressing the Call Waiting key again, or by pressing any idle DN key on the set. If the Display key is pressed before the Call Waiting key, the call waiting party information is displayed.
- Conference
  While in a conference call, the Display (DSP) key can be used to obtain information on other keys. However, the Display key is blocked when the CSD key is active.
- DN key (SL-1 and Meridian Modular telephones)
  While the key is active (established, outgoing ringing) will show the source of the destination. While the key is active but not answered (that is, ringing) will show the source of the originator. While inactive will show the number stored that will be used for the 'last number redial' function (if configured).
- Hot Line
  Hot Type I calls are supported by the Display key feature; pressing the Display key and then the Hot Type I key will show the target DN on the originating station's display.
- Message Waiting
  When the telephone is inactive and the DSP key is pressed, followed by the Message Waiting key, the number stored against the key will be displayed.
- Ring Again party
  When the telephone is inactive and the DSP key is pressed, followed by the Ring Again key, the number stored against the key will be displayed.
- Speed Call number
  To display a stored entry the user presses the Display key and the Speed Call key and dials the list number. The list number cannot be abbreviated.
- Voice Call party
  When the telephone is inactive and the DSP key is pressed, followed by the Voice Call key, the number stored against the key will be displayed.

Operating parameters

Digit Display must be enabled for all console types in LD 15, using the prompt OPT.

Only telephones equipped with a Digit Display module can use this feature.

The Display Time and Display Date key cannot be assigned to key 0.

Feature interactions

Attendant Break-In

During Attendant Break-In, the Attendant Console Digit Display shows the DN of the incoming call and the destination DN until the Attendant extends the incoming call to the destination DN and releases the connection.

Autodial Tandem Transfer

Digit Display allows the automatic display of information relevant to normal call processing if the sets have display capability and the Class of Service is ADD or DDS. When the THF key is pressed the display gets cleared, and pressing the ADL key causes the ADL digits to be displayed. However, no ADL digits will be displayed if no Tone and Digit Switch (TDS)/XCT is available to generate the Dual-tone Multifrequency (DTMF) tones for the ADL digits.

Automatic Redial

Dialed numbers are displayed when the Automatic Redial (ARDL) feature is activated. The calling party can dial digits even though a busy tone indication is given.

Digits dialed while on hold are not displayed.When the calling party accepts a redialed call, the dialed numbers are displayed. If the Display (DSP) key and appropriate RGA key are pressed while a call is on hold, the number redialed is displayed.

China – Flexible Feature Codes - Outgoing Call Barring Enhanced Flexible Feature Codes - Outgoing Call Barring

Meridian 1 proprietary telephones with displays do not display the OCB level and the Station Control Password (SCPW) when OCB FFCs are dialed. This protects the security of the SCPW.

Centralized Multiple Line Emulation

The digit display of the station picking up a parked call recall shows the parked call's access code followed by the parked call's access-identification code. If the picked-up call is a group member call, the display shows the group number of the picked-up station.

Dial Intercom

The digit display will be cleared when the Dial Intercom Group (DIG) key is pressed. When the user dials the DI code, the digits of the code are displayed. When the call is answered, the DI code of the calling party appears on the display of the called party.

If either party presses the Release key or goes on hook during a DIG call, the displays of both parties are cleared. If either party presses the Hold key, the display of the holding station is cleared but the display of the other station remains unchanged. When the held call is reestablished, the holding station redisplays the DIG number of the other party.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The digit display rules for DPNSS1 UDP are based on what is currently done in an MCDN.

Group Hunt

Until a call is answered, the calling party will see the dialed DN. When the call is answered, the caller will see the dialed DN appended with the DN and name, if Calling Party Name Display (CPND) is equipped, of the called party. The terminating set will always see the originating DN appended by a Pilot DN

Hot Line

A Display key on a telephone with a Hot Line appearance will display the Hot Line target DN data stored for that key.

INIT ACD Queue Call Restore

Call information associated with Digit Display is lost after system initialization and call restoration.

LOGIVOX Telephone

During manual dialing or last number redial, the display shows the dialed digits, even if the set has display denied Class of Service. If the set has LOGIVOX denied Class of Service, each digit is shown twice.

Override
Override, Enhanced
Override, Priority

The Digit Display of the telephone being overridden changes to the Directory Number (DN) of the overriding telephone once Priority Override is accomplished.

Pretranslation

The Pretranslation digit is displayed as it was dialed, but if the call is put on hold, the digits of the pretranslated DN are displayed

Feature packaging

Digit Display (DDSP) package 19 has no other feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Configure Digit Display for Attendant Consoles for each customer.
2. LD 11 – Configure Digit Display for Meridian 1 proprietary telephones.
3. LD 12 – Configure Digit Display for each Attendant Console.

LD 15 – Configure Digit Display for Attendant Consoles for each customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Attendant console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(XDP) IDP	(Exclude) include Digit Display capability for Attendant Consoles of this customer.

LD 11 – Configure Digit Display for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

CLS	(NDD) DDS ADD	Telephone is not equipped with a Digit Display. Calling Party information is displayed after call is answered (delayed display source). Calling Party information is displayed during call processing (Automatic Digit Display).
KEY	xx DSP xx TAD	Add a Digit Display key (must be key/lamp pair). Add a Time and Date key (must be key/lamp pair). xx = key number.

LD 12 – Configure Digit Display for each Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DLEN	(8) 16	Digit Display entry length (the default is 8). This prompt applies to QCW consoles only.
KEY	xx DCW xx DDT xx DPD xx DPS xx DTM xx MDT xx MTM	Add display Call Waiting key. Add display Date key. Add display Destination key. Add display Source key. Add display Time key. Add display/change Date key. Add display/change Time key.

Feature operation

No specific operating procedures are required to use this feature.

Digital Private Network Signaling System

British Telecom's Digital Private Signaling System No. 1 (DPNSS1) is the open signaling protocol standard for intelligent private network digital connections. DPNSS1 provides the signaling capability to establish simple telephony and data calls, as well as supplementary features.

The following DPNSS1 features have been introduced:

- DASS2/DPNSS1 – Integrated Digital Access
- DPNSS1 Attendant Call Offer
- DPNSS1 Attendant Timed Reminder Recall and Attendant Three-party Service
- DPNSS1 Call Back when Free and Next Used
- DPNSS1 D-channel Handler Interface Expansion
- DPNSS1/DASS2 to ISDN PRI Gateway
- DPNSS1 Extension Three-party Service
- DPNSS1 Loop Avoidance
- DPNSS1 Redirection
- D-Channel Interface Expansion for DASS2/DPNSS1
- DPNSS1 Route Optimization
- DPNSS1 Step Back on Congestion
- DPNSS1 Executive Intrusion , and
- DPNSS1/DASS2 Uniform Dialing Plan Interworking.

For more information on DPNSS1, see *DPNSS1* (553-3001-372).

Digital Trunk Interface – Commonwealth of Independent States

Contents

This section contains information on the following topics:

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Feature description

The information presented in this section does not pertain to all market regions. Contact your system supplier or your Nortel Networks representative to verify support of this product in your area.

The Commonwealth of Independent States (CIS) Digital Trunk Interface (DTI) feature allows the system to connect to Direct Inward Dialing (DID)/ Central Office Trunk (COT) trunks to a CIS Public Exchange/Central Office and to a CIS toll exchange.

To satisfy the unique requirements of CIS DTI signaling, two new trunk cards have been introduced: a dual 2 Mbps Enhanced Network (ENET) styled digital trunk card (CDTI2); and a 2 Mbps digital trunk card (CSDTI2) for use with Small Systems. The CDTI2 card provides 60 voice or data 64 kbps channels, whereas the CSDTI2 card provides 30 voice or data 64 kbps channels. Each card occupies one card slot on the common equipment shelf (CDTI2), or on the Small System base cabinet (CSDTI2)

In addition to most of the features provided by digital trunks, the CDTI2 and CSDTI2 cards provide the following features intended for the CIS market:

- CIS digital trunk signaling (outgoing, incoming toll, and incoming local calls)
- Automatic Number Identification (ANI) transmission for outgoing calls on request from the Public Exchange
- Special disconnect procedure (two-way release) on incoming local answered calls initiated by the Public Exchange to provide Malicious Call Trace
- Unanswered free special service calls – outgoing calls that remain unanswered are recognized in a special manner to allow the called party (special service operator) to disconnect the calls
- CIS transmission plan
- Downloading the required firmware mode per loop, and
- Dial tone provided internally to the calling party by the system after seizure of an outgoing CIS trunk. However, for outgoing call terminating to a busy, vacant, invalid, or restricted DN, the system does not provide busy/overflow tone. The Public Exchange will send the tone on the speech path.

The CIS DTI trunk provides significant improvement on real-time impact for dial pulse outpulsing and digit collection by transferring these processes from the software to the firmware. The trunk state change validation timing is performed by the firmware. A Firmware Unproductive Timer is used to prevent a call on a CIS trunk from remaining unanswered for too long.

Operating parameters

CDTI2 and CSDTI2 cards do not support Periodic Pulse Metering, continuous pulse detection, or echo suppression.

The only line signaling supported for CIS is a two-bit ABCD protocol.

The data in ANI always refers to the originator of the outgoing call. If the call is transferred, the ANI information is not changed and therefore may be different than that of the set currently involved in the call.

On outgoing toll calls, there is no delay. On outgoing local calls, there is a 700 millisecond delay in the Answer signal recognition before the call is established.

Incoming and outgoing trunks cannot be mixed within the same route.

Toll Operator Break-In/Trunk Offer is not supported.

Toll Operator Manual Ringing is not supported.

MF Shuttle Register Signaling is not supported.

CSDTI2 cards are required for Succession 1000M Cabinet and Meridian 1 Option 11C Cabinet systems, while CDTI2 cards are required for all other machine types.

Only ANI transmission is supported.

Feature interactions

Authorization Code

An extension may refer to an Authorization Code to seize an outgoing CIS DTI trunk. The Authorization Code category is used to build the Automatic Number Identification (ANI) message. Thus, a set having a CIS restricting call category can complete a call to the Public Network using the Authorization Code.

Automatic Trunk Maintenance

This feature is not supported on CSDTI2 due to the absence of tone detectors on the Succession 1000M Cabinet and Meridian 1 Option 11C Cabinet.

Called Party Disconnect Control

This feature may not be used in the CIS market because of its signaling requirements.

Computer to circuit-switched network interface

Computer to circuit switched network Interface is not supported on CDTI2/ CSDTI2 because the protocol conversion is not supported.

Call Detail Recording

If ANI is requested to be output in the Call Detail Recording (CDR) record, it will not refer to the CIS DTI2 ANI.

Data Transmission

All features connected with Data Transmission must be used with caution, because the ANI interaction can happen at any time during an outgoing call, thus destroying the transmitted data and disrupting the call.

Dial Tone Detection

Dial tone detection is supported, but with the limitation of low reliability of the tone provided by the Public Exchange.

Incoming Digit Conversion Enhancement
Incoming DID Digit Conversion

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the Direct Inward Dialing (DID) number to dial to reach the set. Therefore, if an external party uses a DN, delivered in an ANI message, for making a call to the corresponding extension, the call may fail.

Japan DTI2

All features related to Japan DTI2 may not be used, because the proper Scan and Signaling Distributor (SSD) messages are not supported in the CDTI2/ CSDTI2 firmware.

Multiple Appearance Directory Number

Since the ANI category is defined on a per set basis, two stations with the same Multiple Appearance Directory Number (MADN) can be assigned different ANI categories.

Periodic Pulse Metering

Periodic Pulse Metering is not supported.

Pulsed E&M DTI2 Signaling

Pulsed E&M is not supported.

R2MFC Calling Number Identification

The category (CAC) used to build the R2MFC Calling Number Identification (CNI) for the analog, digital, and Basic Rate Interface (BRI) sets is also used to build the CIS Automatic Number Identification (ANI). The meaning of CAC is different between the R2MFC CNI signalization and the CIS signalization (analog BRI and digital). R2MFC CAC prompt values are in the range of 0-10, with the default value of 0. CIS CAC prompt values are in the range of 0-9, with the default value of 3.

If the MFC package is equipped, but not the CIST package, the CAC prompt uses the R2MFC range and default. If the CIST package is equipped, whether or not the MFC package is equipped, the CAC prompt uses the CIS range and default.

Special Dial Tones after Dialed Numbers

Special Dial Tones can be used to provide dial tone after the system user has dialed the digit “9” (Local Exchange access code).

Tandem Switching

If an ISDN TIE incoming trunk (MCDN, QSIG, DPNSS1) with Calling Line Identification (CLID) and Originating Line Identification (OLI) available seizes the CIS DTI2 outgoing trunk, the ANI DN to be used for sending to the CIS Public Exchange is extracted from this CLID/OLI.

In any other case, the ANI sent to the CIS Public Exchange is based on the local system node (that is, tandem node) definitions.

Virtual Network Services

Virtual Network Services via CIS DTI2 is not supported.

Feature packaging

This feature is packaged as Commonwealth of Independent States Trunk Interface (CIST) package 221.

The following packages are required:

- Flexible Tones and Cadences (FTC) package 125
- 2 Mbps Digital Trunk Interface (DTI2) package 129
- International Supplementary Features (SUPP) package 131
- Flexible Numbering Plan (FNP) package 160
- Meridian 1 Intelligent Peripheral Equipment (XPE) package 203
- Meridian 1 Extended Conference, TDS and MFS (XCT0) package 204, and
- Meridian 1 Superloop Administration (XCT1) package 205.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Change system configuration data.

2 LD 73 – Define DTI2 data.

3 LD 73 – Define the SICA table for CDTI2/CSDTI2.

4 LD 97 – Define dial pulse make-break ratio.

5 LD 15 – Define busy tone/overflow tone time out.

6 LD 16 – Add or change route data for CIS DTI2 trunks.

7 LD 16 – Add or change route data for an incoming, non CIS DTI2, trunk.

8 LD 14 – Add or change trunk data for CIS DTI2 incoming and outgoing trunk.

9 LD 10 – Specify ANI category for CIS DTI2 calls.

10 LD 11 – Specify ANI category for CIS DTI2 calls.

11 LD 12 – Specify ANI category for CIS DTI2 calls.

12 LD 27 – Add or change Digital Subscriber Loop (BRI set) for CIS.

13 LD 88 – Add or change the Authcode data block.

14 LD 56 – Configure the dial tone.

15 LD 56 – Configure Tone to Last Party.

16 LD 18 – Add or change Speed Call lists, System Speed Call lists, Group Call lists, Enhanced Hot Line lists, Pretranslation lists, and Special Service lists. Special Service lists can now handle the Special Service Unanswered Call (SSUC) call type.

LD 17 – Change system configuration data.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters
...		
- PCML	(MU) A	System Companding Law.
TYPE	CEQU	Common Equipment
...		
- DTI2	<loop> <loop>...	Define CDTI2/CSDTI2 loops exactly like existing DTI2/ SDC2.

LD 73 – Define DTI2 data.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	DTI2 Data Block.
FEAT	LPTI	Loop timers and some other per DTI2 loop defined parameters.
LOOP	<loop>	DTI2 loop number.
CDTI2	YES	CDTI2/CSDTI2 card.
P DIGIT (S)	PXXX	DP outpulsing will be sent on signaling bit A.
P METR (R)	NO	Pulse Metering.
SASU	1024	Seize Acknowledgment Supervision timer is defined in milliseconds (rounded to the closest multiple of 128 milliseconds).
MFAO	YES	Multi-frame alignment option used.
SZNI	NO	Seize Not Idle option not used.
LCLB	NO	Lockout Clear Back option (send CLR-BK signal to DID in lockout) not used.
UCFS	1101	Unequipped Channel Fault Signal – ABCD bits to be sent on unequipped channel. The default value of 1101 is acceptable.
MFF	(AFF) CRC	Alternate Frame Format or CRC4 may be chosen.
...		
FRFW	NO	Prompted only if French Type Approval (FRTA) package 197 is equipped.
CISFW	YES	Prompted only if Commonwealth of Independent States – Digital Trunk Interface (CIST) package 221 is equipped and CDTI2 = YES.

LD 73 – Define the SICA table for CDTI2/CSDTI2.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	DTI2 Data Block.
FEAT	ABCD	Signaling category.
SICA	2-16	SICA table number.
...		
Incoming/ Outgoing Calls		
IDLE(S)	0101, 1101	Idle on backward sent, where: 0101 = incoming trunks (local and toll), and 1101 = outgoing trunks.
IDLE(R)	0101, 1101	Idle on backward sent, where: 0101 = incoming trunks (local and toll), and 1101 = outgoing trunks.
FALT(S)	1101	Fault (referred to as blocked in CCITT terminology).
FALT(R)	1101	Fault (referred to as blocked in CCITT terminology).
Incoming Calls		
E_SEZ(R)	1001	Seize.
SEZD(R)	NO	Seize for voice calls.
P CALL (R)	NO	Signal sent during seize by an incoming CO trunk.
SEZA (S)	1101	Seize Acknowledge.
- TIME	150	Time in milliseconds.
P DIGT(R)	Pxxx	DP Digits received decadic pulses.
NRCV(S)	NO	Number received.

P EOSF(S)	NO	Pulsed End of Selection Free is not used.
EOSF(S)	NO 1001	Steady End of Selection Free, where: NO = local trunk, and 1001 = incoming toll trunk.
P EOSB(S)	NO	Pulsed End of Selection Busy is not used.
EOSB (S)	0001	Steady End of Selection Busy.
P OPCA(R)	NO	Operator calling.
E_CONN(S)	1001 1101	Connect Send (Answer), where: 1001 = local trunk, and 1101 = incoming toll trunk.
CONN(R)	1001 0001	Connect received, where: 1001 = local trunk, and 0001 = incoming toll trunk.
P RRC(S)	NO	Register recall.
P BURS(S)	NO	Bring up receiver for L1 networking.
P BURS(R)	NO	Bring up receiver for L1 networking.
CLRB(S)	0001 1001	Clear Back (B Ring Off), where: 0001 = local trunk, and 1001 = incoming toll trunk.
CLRF(R)	0001 NO	Clear Forward (A Ring Off), where: 0001 = local trunk (used only to start two-way release), and NO = incoming toll trunk.
P OPRS(R)	NO	Operator manual recall.
P NXFR(S)	NO	Network transfer.
P ESNW(S)	NO	ESN wink.
P CAS(S)	NO	Centralized attendant.
Outgoing Calls		

E_SEZ(S)	1001	Seize.
SEZD(S)	NO	Seize for data calls.
SEZA(R)	1101	Seize Acknowledge.
P WNKS(R)	NO	Wink start.
P EOS(R)	NO	End of selection busy.
E_CONN(S)	NO	Connect.
E_CONN(R)	1001	Connect Receive (Answer).
P OPRC(R)	NO	Operator recall for special services.
P BURS(S)	NO	Bring up receiver for L1 networking.
P BURS(R)	NO	Bring up receiver for L1 networking.
CLRB(R)	0001	Clear Back (B Ring Off).
CLRF(S)	NO	Clear Forward (the same as the IDLE(S) signal).
P NXFR(R)	NO	Network transfer.
P ESNW(R)	NO	ESN wink.
P CAS(R)	NO	Centralized Attendant Service.

LD 97 – Define dial pulse make-break ratio.

Prompt	**Response**	**Description**
REQ	CHG	Change.
TYPE	SYSP	System parameters.
INTN	YES	A-law should be used as system companding law.
...		
P10R	(50)-70	Make-break ratio for primary 10 pulses per second dial pulse dialing.

P12R	(50)-70	Make-break ratio for secondary 10 pulses per second dial pulse dialing.
P20R	(50)-70	Make-break ratio for 20 pulses per second dialing.

LD 15 – Define busy tone/overflow tone time out.

Prompt	Response	Description
REQ:	NEW CHG	New or change.
TYPE:	TIM	Gate opener.
...		
- BOTO	30	Busy tone/overflow tone time out (in seconds).

LD 16 – Add or change route data for CIS DTI2 trunks.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID COT	Trunk type, where: DID = incoming trunks route, and COT = outgoing trunks route.
...		
DTRK	YES	Digital trunk.
DGTP	DTI2	Digital trunk type for route.
...		
ICOG	ICT OGT	Incoming trunk. Outgoing trunk.
...		

CNTL	YES	Changes to controls or timers.
TIMR	DDL 0	Delay Dial Timer not needed.
TIMR	DSI 49992	Disconnect supervision timer (five-second value, rounded to the nearest 128 ms.).
TIMR	EOD 13952	End of dial timer (default value).
TIMR	SFB 25	Seize Fail Busy timer. The recommended value for trunks with seizure supervision is 25 seconds.
TIMR	GTI 0	Incoming Guard timer must be defined equal to zero. Incoming CIS DTI2 trunks only. For CIS DTI2 trunks no guard timing is necessary on the incoming side. Immediately after sending the ”IDLE” signal, the incoming trunk may be reseized by the CO.
TIMR	ATO 128-(4992)-65408	ANI time out timer in milliseconds. For CIS outgoing trunk routes this defines the time delay performed after the outpulsing of the toll access code. During this delay further outpulsing is temporarily halted until the special message from the card firmware confirms that a successful ANI request/response interaction has been performed.
...		
NEDC	ORG ETH	Near end disconnect control, where: ORG = originating end disconnect control for incoming calls, and ETH = either end control for outgoing calls.
FEDC	ORG ETH	Far end disconnect control, where: ORG = originating end disconnect control for incoming calls, and ETH = either end control for outgoing calls.
CDPC	NO	The system is not the only controlling party on incoming calls.
...		
OPCB	NO	External operator features not allowed on this route.
...		

CGPC	NO	Calling party control of calls not enabled.
CDCT	NO	Called party control of call is not enabled.
DDO	NO	Do not delay digit outpulsing for DOD trunks.
...		
DTD	NO	Dial tone detection is not to be performed on this route.
...		
CDR	YES	CDR to output for calls on trunks in this route.
...		
OAL	YES	CDR on all outgoing calls.
...		
OAN	NO	CDR on answered outgoing calls. It is not used because of free special service calls, which are not answered.
NATL	NO	North American toll scheme is not used.
TDG	8	Toll digit (list of digits after the trunk access code which indicate toll calls). This can also be defined in LD 18.
...		
PRDL	YES	Partial dial timing is equipped using EOD.
DNSZ	(0)-7	Number of digits expected on DID routes. 0 (the default) indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Duration of busy/overflow tone to be returned on DID route in seconds.
...		
LEC	0-9999999	Local Exchange Code.

ADDG	0-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.

LD 16 – Add or change route data for an incoming, non CIS DTI2, trunk.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	RDB	Route Data Block.
...		
ICOG	ICT IAO	Incoming trunk. Incoming and outgoing trunk.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Remote DN Length.

Note: This trunk may be any kind of trunk. If this trunk, used as an incoming trunk, originates an outgoing call to a CIS DTI2 trunk, its CAC and ANDN are used in the ANI information sent out.

LD 14 – Add or change trunk data for CIS DTI2 incoming and outgoing trunk.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	DID COT	Direct Inward Dialing (for incoming trunks), or Central Office Trunk (for outgoing trunks).
...		
SICA	2-16	Signaling category table number. ***Note:*** standard default SICA table (number 1) may not be used for CIS DTI2 trunks. CIS DTI2 trunks for incoming local and incoming toll calls must have different SICA tables.
PDCA	(1)-16	PAD table number.
PCML	A	Only A-law companding may be used on the CIS DTI2 trunk line.
...		
CIST	(NO) YES	This prompt appears for incoming trunks only (ICOG = ICT in LD 16), where: YES = toll trunk, and NO = local trunk.
...		
CLS	(DIPF DIP	Dial pulse execution, where: DIP = outpulsing by firmware, digit collection – traditional, by software, and DIPF = outpulsing and digit collection are performed by firmware.
	(P10) P12	Make-break ratio for dial pulse dialing.

LD 10 – Specify ANI category for CIS DTI2 calls.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	500	500/2500 telephone data block.
...		
CAC	0-(3)-9	Specify ANI category for CIS DTI2 calls.
CLS	(DNAA) DNAD	DN of set (allowed) not allowed for use in ANI messages.

LD 11 – Specify ANI category for CIS DTI2 calls.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	aaaa	Telephone data block, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CAC	0-(3)-9	Specify ANI category for CIS DTI2 calls.
CLS	(DNAA) DNAD	DN of set (allowed) not allowed for use in ANI messages.

LD 12 – Specify ANI category for CIS DTI2 calls.

Prompt	Response	Description
REQ	NEW CHG	Add or change.
TYPE	ATT 1250 2250	Type of Attendant Console.
...		
DNAN	(DNAA) DNAD	For CIS ANI purposes, the ANI DN will be LDN0 (defined in LD 15).

LD 27 – Add or change Digital Subscriber Loop (BRI set) for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add or change.
TYPE	DSL	Digital Subscriber Loop.
...		
CAC	0-(3)-9	Specify ANI category for CIS DTI2 calls.
CLS	(DNAA) DNAD	DN of set (allowed) not allowed for use in ANI messages.

LD 88 – Add or change the Authcode data block.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	AUB	Authcode data block.
...		
CLAS	(0)-115	Classcode value assigned to Authcode (NAUT).
...		
NCOS	(0)-99	Network Class of Service group number.
CAC	0-(3)-9	ANI category for CIS DTI2 calls.

LD 56 – Configure the dial tone.

Prompt	Response	Description
REQ	NEW CHG	New or change.
TYPE	DTAD	Special dial tone after dialed number.
DDGT	9	Use “9” as the outgoing local access code.

TONE	SRC1	Dial tone to be provided after the dialed digit 9 (Source Tone 1).
...		
REQ	NEW CHG	New or change.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	FTC table number.
DFLT	0-31	Default FTC table.
...		
SRC	YES	Change Source Tones (SRC1-SCR8).
SRC1		Source Tone 1.
TDSH	0 0 0 3	Tone number 3 on QPC609 provides 400 Hz, -23 db.
XTON	159	NT8D17 TDS tone code: 420 Hz, -25 db, A-law.
XCAD	0	NT8D17 cadence code for FCAD (steady tone).

LD 56 – Configure Tone to Last Party.

Prompt	**Response**	**Description**
REQ	NEW CHG	New or change.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	FTC table number.
DFLT	0-31	Default FTC table.
RING	<CR>	
HCCT	YES	Change the TDS card controlled cadence tones.
...		
TLP		Tone to Last Party.

TDSH	0 0 31 3	Cadence 31 in MCAD table will provide repeating 256 ms burst and 256 ms silence. Tone number 3 on QPC609 provides 440 Hz, -23 db.
XTON	159	NT8D17 TDS tone code: 420 Hz, -25 db, A-law.
XCAD	31	NT8D17 cadence code for FCAD.
TLTP	30	Tone to Last Party timer in seconds.
...		
REQ	NEW CHG	New or change.
TYPE	MCAD	Master Cadence table.
WCAD	31	Cadence number.
CDNC	0051 0051	Repeating 256 ms burst and 256 ms silence.
...		
REQ	NEW CHG	New or change.
TYPE	FCAD	Firmware Cadence table.
WCAD	31	Cadence number.
CDNC	0060 0060	Repeating 300 ms burst and 300 ms silence.

LD 18 – Add or change Speed Call lists, System Speed Call lists, Group Call lists, Enhanced Hot Line lists, Pretranslation lists, and Special Service lists. Special Service lists can now handle the Special Service Unanswered Call (SSUC) call type.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	SSL	Special Service List.
SSL	1-15	SSL number.
SSDG	0-999	Special Service Digits combination.

CDPC	(NO) YES	Called Party Control mark.
TOLL	(NO) YES	Toll access code mark.
ALRM	(NO) YES	Alarm digits mark.
TNDM	(NO) YES	Tandem mark. Send MFC “H” tandem signal.
SSUC	(NO) YES	Special Service Unanswered Call mark. If the outgoing call is recognized as SSUC (first 1-4 digits outpulsed to the trunk are equal to the SSDG with SSUC = YES), then such a call requires some specific disconnect treatment.

Feature operation

No specific operating procedures are required to use this feature.

Digitone Receiver Enhancements

Contents

This section contains information on the following topics:

Feature description 137
Operating parameters 138
Feature interactions 138
Feature packaging 138
Feature implementation 138
Feature operation 139

Feature description

The Digitone Receiver Enhancements feature consists of the Digitone Receiver Time out Enhancement and the Quadruple Density Digitone Receiver Card.

An enhancement to Digitone receiver (DTR) time out prevents the situation in which the far-end of an outgoing call from a Dual-tone Multifrequency (DTMF) telephone or trunk is answered before speechpath can be established.

This problem can occur when trunks without answer supervision are used, and the called party answers quickly. Without answer supervision, the speech path is established upon time out of the end-of-dialing timer. It is possible for the far-end station to answer before this time out.

The timer enhancement will prevent this situation from occurring by holding back outpulsing of the last digit until a half-second before end-of-dialing time out. This leaves only a half-second interval in which the far-end station could answer before speechpath is established.

This DTR timer enhancement applies to DTRs of all densities, and for all trunk calls made from DTMF telephones or trunks, except for:

- MFC or MFE calls
- terminating trunks that have answer supervision
- Electronic Switched Network (ESN) calls

Operating parameters

This feature is not supported on the 1.5 Mbit Digital Trunk Interface (DTI).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 13 – Create or modify data blocks for Digitone Receivers.

Prompt	Response	Description
...		
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	4D	Enter 4D if the unit is on a quadruple density circuit pack (not allowed if the network loop is not configured for quadruple density).

Feature operation

No specific operating procedures are required to use this feature.

Direct Inward Dialing Call Forward No Answer Timer

Contents

This section contains information on the following topics:

Feature description 141
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Feature packaging 143
Feature implementation 143
Feature operation 143

Feature description

This feature introduces the Direct Inward Dialing Call Forward No Answer (DFNR) timer that, when expired, allows an unanswered Direct Inward Dialing (DID) call to be routed to the attendant after the last stage of Call Forward No Answer (CFNA) or hunt treatment has been completed (the maximum number of CFNA or hunt steps is two). The DFNR timer is customer-defined as a number of ring cycles in LD 15.

The operation of the DFNR option can be overridden or disabled, depending on the definition of the FNAD option in LD 15. If FNAD has been defined as attendant (ATT), the DFNR timer is overridden, since unanswered DID calls are automatically routed to the attendant. If FNAD has been defined as NO, DFNR is disabled. If FNAD has been defined as Hunt (HNT) or Forward DN (FDN), the DFNR timer is applied after the last stage of CFNA or hunt treatment has been completed.

Operating parameters

DFNR does not apply to Automatic Call Distribution (ACD) calls, nor does it apply to non-DID calls.

The DFNR overrides the Forward Number Allowed (FNA) or Forward Number Denied (FND) Class of Service of the called party.

Feature interactions

Attendant Recall

The Direct Inward Dialing Call Forward No Answer Timer does not apply to an answered DID call that is extended to an unanswered station by the attendant – the call is recalled to the attendant using the Attendant Recall feature.

Call Forward No Answer Hunting

Call Forward No Answer and Hunting take precedence over the Message Center feature.

Call Waiting Redirection

The Direct Inward Dialing Call Forward No Answer Timer is applied after the last stage of Call Forward No Answer or SFNA treatment resulting from the Call Waiting Redirection feature for DID Call Waiting calls.

Feature packaging

The Direct Inward Dialing Call Forward No Answer (DFNR) feature requires the Flexible Feature Codes (FFC) package 139.

Feature implementation

LD 15 – Define the Number of Ring Cycles.

Prompt	Response	Description
REQ:	NEW CHG	Add. Change.
TYPE:	RDR	Call Redirection
...		
- DFNR	(0)-15	DID Forward No Answer Ring cycles, prompted if the FNAD prompt is not set to ATT or NO. Defines the number of ringing cycles before a DID call is Slow Answer recalled to the Attendant Console after the last stage of CFNA or Hunt treatment has been completed (the maximum number of CFNA or hunt steps is two). If DFNR = 0 then DID CFNA is disabled

Feature operation

No specific operating procedures are required to use this feature.

Direct Inward Dialing Recall Features on DTI2 for Italy

Contents

This section contains information on the following topics:

Feature description 145
Operating parameters 146
Feature interactions 147
Feature packaging 147
Feature implementation 147
Feature operation 150

Feature description

Direct Inward Dialing (DID) Recall Features on DTI2 for Italy consists of DID Offering and DID Recall.

DID Offering

When a DID call placed on a DTI2 trunk terminates on a busy set, the system replies by sending an End of Selection Busy (EOSB) signal on the calling channel to inform the Public Exchange/Central Office that no further call modification will be performed. Busy tone is returned while waiting for the release signal from the Central Office (IDLE). The new DID Offering feature enables the external Central Office operator to reroute the call to the attendant by sending the Operator Recall Signal (OPRS) instead of the IDLE signal. Upon receipt of the OPRS signal, the call is presented to the Attendant Console on the Recall (RLL) Incoming Call Indicator (ICI) key.

DID Recall

When an established DID call placed on a DTI2 trunk is released by called party (internal set), the system sends a Clear Backward (CLRB) signal on the calling channel to inform the Central Office that the call has been disconnected. Upon receipt of this signal, the Central Office should reply with the IDLE signal to confirm the disconnection of the call. At this point, the new DID recall feature allows the external Central Office operator to reroute the call to the Attendant Console by sending the OPRS signal instead of IDLE. The system will detect the OPRS as a valid signal and the call will be presented to the Attendant Console on the RLL ICI key.

Operating parameters

Both DID Offering and DID Recall currently only support Type Approval in Italy and are not commercially available.

The QPC536 Digital Trunk Interface and NTAK10 (XDTI) cards are required.

This feature only works on DTI2 trunks.

Feature interactions

Basic Rate Interface (BRI) Special Call Forward Busy

This feature takes precedence over the DID offering; when the conditions for the BRI Special Call Forward Busy are met, the call is diverted to the Attendant Console without waiting for the OPRS signal. When the BRI Special Call Forward Busy feature fails or is not enabled, busy tone is returned to the Central Office and the DID offering can be activated.

Forward Busy

The DID offering is available only after the End of Selection Busy signal has been sent by the Central Office. This signal is provided to the Central Office trunk only if the busy set is configured with Forward Busy Denied (FBD) Class of Service.

Network Attendant Services (NAS)

Incoming DID calls which are Offered or Recalled to the attendant may receive NAS treatment. This feature requires no modification.

Feature packaging

Direct Inward Dialing (DID) Recall Features on DTI2 for Italy are included in the existing 2 Mbit Digital Trunk Interface (DTI2) package 129, which requires International Supplementary Features (SUPP) package 131.

Feature implementation

DID Offering

Task summary list

The following is a summary of the tasks in this section:

1 LDs 10/11 –Set the Class of Service to FBD.

2 LD 16 – Set DID Recall for this Rate.

3 LD 73 – Configure the SICA table for the DID Offering feature.

4 LD 16 – Set DID Recall to Attendant for this rate.

5 LD 73 – Configure the SICA table for the DID Recall feature.

LDs 10/11 –Set the Class of Service to FBD.

Prompt	Response	Description
REQ:	NEW CHG	New or change.
TYPE:	aaaa	Telephone type, where aaaa = 500, SL-1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	FBD	Forward Busy Denied.

LD 16 – Set DID Recall for this Rate.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
...		
RCAL	DRA	Set DID Recall to ATTN for this route.

LD 73 – Configure the SICA table for the DID Offering feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	2 Mbit DTI.
FEAT	ABCD	Digital signaling category table.

SICA	nn	Signaling Category table number.
INCOMING/OUTGOING CALLS		
IDLE(R)	ABCD	(Receive) IDLE signal bits.
INCOMING CALLS		
P EOSB(S)	ABCD	End of Selection Busy (receive) signal.
- TIME	(100)-150	Duration of the EOSB(S) signal in milliseconds.
...		
P OPRS(R)	ABCD	Operator (receive) recall signal.
- TIME	xxxx yyyy	Time for OPRS(R) in milliseconds, where: xxxx = 8-(48)-2040, and yyyy = xxxx-(128)-2040.

DID Recall

LD 16 – Set DID Recall to Attendant for this rate.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route Data Block.
...		
RCAL	DRA	Set DID Recall to attendant for this route.

LD 73 – Configure the SICA table for the DID Recall feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	2 Mbit DTI.
FEAT	ABCD	Digital signaling category table.
SICA	nn	Sica table number.
...		
INCOMING CALLS		
CLRB(S)	ABCD	Clear Backward (send) signal.
...		
P OPRS(R)	ABCD	Operator (receive) recall signal.
- TIME	xxxx yyyy	Time for OPRS(R) in milliseconds, where: xxxx = 8-(48)-2040, and yyyy = xxxx-(128)-2040.

Feature operation

No specific operating procedures are required to use this feature.

Direct Inward Dialing to TIE Connection

Contents

This section contains information on the following topics:

Feature description 151
Operating parameters 152
Feature interactions 152
Feature packaging 152
Feature implementation 152
Feature operation 153

Feature description

This feature allows DID-to-TIE connections, subject to all trunk barring, Trunk Group Access Restrictions (TGAR), Trunk Access Restriction Groups (TARG), and other Class of Service restrictions. When the end-of-dialing timer detects that end-of-dialing is reached for an outgoing TIE trunk the Call Forward No Answer (CFNA) timer is started.

If the CFNA timer expires prior to detecting an answer signal the call is intercepted to the attendant. If a routed call receives a busy signal from an extension, the busy signal is returned to the DID. If the DID does not go on-hook before the CFNA recall timer expires, the call is routed to the attendant.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the DID-to-TIE Connection feature described above.

The DID-to-TIE Connection feature is not available on 1.5 Mbps digital, Japanese DMI, PRI2 or DPNSS trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Allow DID-to-TIE connections.

2 LD 16 – Define the Number of digits expected on DID rate.

LD 15 – Allow DID-to-TIE connections.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	NET	ISDN and ESN networking options.
...		
- DITI	(NO) YES	DID-to-TIE connections (are not) are allowed.

LD 16 – Define the Number of digits expected on DID rate.

Prompt	Response	Description
...		
DNSZ	(0)-7	Number of digits expected on DID route. 0 indicates no fixed number.

Feature operation

No specific operating procedures are required to use this feature.

Direct Inward System Access

Contents

This section contains information on the following topics:

Feature description 155
Operating parameters 156
Feature interactions 157
Feature packaging 159
Feature implementation 160
Feature operation 162

Feature description

Direct Inward System Access (DISA) allows selected users to access the system from the public or private network by dialing a special Directory Number (DN) assigned by the customer. The number can be dialed from any Digitone telephone outside the network. Once the Direct Inward System Access (DISA) call has been answered, the user can access any of the following features and capabilities offered through Direct Inward System Access:

- Calls to any station within the customer group
- Trunk calls (such as calls to a Public Exchange/Central Office, a TIE trunk, or paging and dictation trunks)
- Basic/Network Authorization Code (BAUT/NAUT)

- Call Detail Recording (CDR) and Call Detail Recording Charge Account, and
- Basic/Network Alternate Route Selection (BARS/NARS) and Automatic Number Identification (ANI) route selection.

Each special Directory Number (DN) dialed by a DISA user is associated with a particular DISA Directory Number. Any number of DISA DNs can be assigned, provided that they are consistent with the numbering plan of the customer. Access rights are determined by the Class of Service and Trunk Group Access Restrictions (TGAR) associated with the DISA number. Calls to DISA can be placed on dedicated, auto-terminate incoming trunks (Central Office [CO], Foreign Exchange [FX], or Wide Area Telephone Service [WATS]) and TIE or Direct Inward Dialing (DID) trunks, all of which must have proper supervision.

As a safeguard against unauthorized use, an authorization code or special security code of one to eight digits can be assigned for each DISA DN. The security code must be entered before any system resources can be used. Additionally, a secure data password can be provided to enable the customer to create, modify, or remove information concerning DISA.

Operating parameters

The features not available to DISA users are those that require a switchhook flash (such as Call Transfer, Conference, Hold, or Ring Again). Also unavailable are features requiring that predefined data be assigned for the DN (for example, Speed Call), and other features that are not applicable to DISA calls (such as Call Pickup and Call Forward).

Any CO, FX, or WATS trunk route can be designated as an auto-terminate route, allowing incoming calls in the route to terminate on one particular DN rather than going to the attendant. Several trunks can specify the same DISA DN, or each trunk can specify a different DISA DN.

Only trunks that give disconnect supervision can be used to provide access to DISA. Therefore, trunks dedicated to DISA (CO, FX, or WATS) must have a ground start signaling arrangement. Incoming DISA calls on trunks without disconnect supervision will not be allowed. For these calls, overflow tone is given to TIE, DID, and Common Controlled Switching Arrangement (CCSA) trunk calls, and calls on CO, FX, and WATS trunks are intercepted to the attendant.

Trunks dedicated to DISA may also be used as normal outgoing trunks.

Feature interactions

Access Restrictions

Access restrictions are assigned to the DISA DN as they are to any station within the system. Separate access restrictions are also assigned to authorization codes used by DISA callers.

Attendant Busy Verify
Busy Verify

Attendant Busy Verify applies only to DNs within the system. If an attendant tries to use the feature to enter a DISA DN, overflow tone is returned.

Basic/Network Alternate Route Selection (BARS/NARS)

The BARS/NARS features function on a DISA call as if it had been originated from inside the system.

Basic/Network Authorization Code (BAUT/NAUT)

This feature can be used in conjunction with DISA to allow a user access to more resources than are normally available. The Authorization Code must be entered, in addition to the security code (if required), using the applicable Special Prefix (SPRE) code followed by the authorization access code 6, or by an applicable Flexible Feature Code. If authorization codes are required, a valid Authorization Code must be entered after the DISA security code (no SPRE code is needed).

Call Forward/Hunt Override Via Flexible Feature Code

DISA is not supported. Any attempt to dial the Call Forward/Hunt Override Via Flexible Feature Code will be ignored and access denied treatment will be returned.

Call Detail Recording

If the customer and trunk route on which the incoming DISA call is being made have the applicable Call Detail Recording (CDR) options in effect, particulars of the call are recorded when it is established. There is no special indication on the CDR record that this was a DISA call. If the incoming trunk route is not specified for CDR options, recording depends on what has been specified by the customer for any outgoing trunks seized by the DISA caller.

China Number 1 Signaling - Called Party Control

If an external station is allowed access to the trunk on which a Special Service resides using Direct Inward System Access (DISA), the station may also access that Special Service. However, Called Party Control is not supported.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

DISA is not supported in a DPNSS1 UDP network.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The Electronic Lock feature cannot be activated or deactivated when accessing the node through DISA.

Generic XFCOT Software Support

This feature allows selected external users to access the system switch by dialing a special directory number, and to use some features of the system as an internal station.

A Direct Inward System Access (DISA) call is allowed on a disconnect supervised or unsupervised loopstart trunk. If a caller on an unsupervised loopstart trunk disconnects during a DISA operation, it is detected by a dial time out or when the call is answered.

Caller disconnection during a DISA operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the operation can then be ended.

ISDN QSIG/EuroISDN Call Completion

Call Completion on Busy Subscriber (CCBS) and Call Completion No Response (CCNR) are not supported on Direct Inward System Access (DISA) calls when the call destination is busy.

Line Lockout
Flexible Line Lockout

The defined Flexible Line Lockout treatment is provided to DISA calls.

New Flexible Code Restriction

If the Direct Inward System Access (DISA) DN has a TLD, CUN, or CTD Class of Service, calls made through DISA are eligible for NFCR treatment.

Night Service Enhancements

It is not possible to assign a Night Service Group Number to any trunk that is a member of a route that is set to auto-terminate on a DISA DN.

Pretranslation

Direct Inward System Access calls are automatically assigned XLST 0.

Scheduled Access Restrictions

Direct Inward System Access (DISA) numbers are not assigned to Scheduled Access Restrictions (SAR) groups and therefore are not affected by SAR schedules.

DISA can be used to manually modify the SAR schedule, provided that the correct FFC and Authorization Code are dialed.

Feature packaging

Direct Inward System Access (DISA) is package 22 and has no other feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 24 – Configure the Direct Inward System Access feature for a customer.
2. LD 16 – Define an auto-terminate trunk route for Direct Inward System Access.
3. LD 14 – Define Direct Inward System Access DNs for trunks in an auto-terminate trunk route.

LD 24 – Configure the Direct Inward System Access feature for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DIS	DISA data.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	System secure data password (0001-9999) allows modifications to the DISA data block. 0000 = disable the password (see LD 15).
DN	xxx...x	DN for DISA access.
SCOD	X xx...xx	DISA security code (1-8 digits). X = remove security code.
AUTR	(NO) YES	Authorization Code is not or is required.
TGAR	xx	Trunk Group Access Restriction to be applied to calls made using DISA (0-15). TGAR can be from 0 to 31.

NCOS	xx	Network Class of Service to be applied to DISA calls.
COS		Class of Service to be applied to DISA calls.
	UNR	Unrestricted.
	CUN	Conditionally unrestricted.
	SRE	Semi-restricted.
	TLD	Toll restricted.
	CTD	Conditionally toll restricted.
	FRE	Fully restricted.
	FR1	Fully restricted 1.
	FR2	Fully restricted 2

LD 16 – Define an auto-terminate trunk route for Direct Inward System Access.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	For Large Systems
	0-127	For Small Systems and Succession 1000 systems
TKTP	aaa	Trunk type.
AUTO	(NO) YES	Route is not or is arranged to auto-terminate incoming calls on the DISA DN.
ICOG	IAO ICT OGT	Incoming and outgoing trunk.
ACOD	xxxx	Trunk route access code.

LD 14 – Define Direct Inward System Access DNs for trunks in an auto-terminate trunk route.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	COT FEX WAT	Trunk type.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XUT	Universal trunk card (prompted for Superloops).
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
ATDN	xxx...x	DISA DN on which incoming calls are to auto-terminate.
SIGL	GRD	Ground Start signaling.

Feature operation

To dial into the system from the public network:

1 Dial the DISA number. You hear a dial tone.

2 Dial the security code, if required.

3 Dial the Authorization Code, if required.

Direct Inward System Access on Unsupervised Trunks

Contents

This section contains information on the following topics:

Feature description 163
Operating parameters 163
Feature interactions 164
Feature packaging 164
Feature implementation 164
Feature operation 164

Feature description

With this enhancement, Direct Inward System Access (DISA) is allowed on Public Exchange/Central Office (CO), FEX, and WATS trunks without disconnect supervision. Without the enhancement, DISA calls on these trunks are intercepted to the attendant. The Timed Forced Disconnect Timer is used to prevent the permanent seizure of the Central Office trunk in cases where the far-end goes on-hook first.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Direct Inward System Access (DISA) package 22.

Feature implementation

LD 16 – Configure the Timed Forced Disconnect Timer.

Prompt	Response	Description
...		
MFC	(NO) YES	Respond with YES to enable Multifrequency Compelled Signaling.
- TIMR	TFD (0)-3600	Timed Force Disconnect in 30-second increments.

Note: In addition, make sure the DISA feature is configured as described in the DISA feature description contained within this document.

Feature operation

To dial into the system from the public network:

1 Dial the DISA number. You hear a dial tone.

2 Dial the security code, if required.

3 Dial the Authorization Code, if required.

Direct Private Network Access

Contents

This section contains information on the following topics:

Feature description 165
Operating parameters 166
Feature interactions 167
Feature packaging 168
Feature implementation 169
Feature operation 171

Feature description

The Direct Private Network Access feature provides enhancements to the processing of Direct Inward System Access (DISA) and Authcode Last Request calls. This feature complements system capabilities to provide an arrangement suitable for long distance resellers. Typically, subscribers to these resellers' services dial in through a DISA port and require some automated digit manipulation, recorded announcements and Authcodes for billing purposes. This feature offers the following capabilities:

DISA Digit Insertion

Once a DISA Directory Number (DN) is accessed, the system automatically inserts from 1 to 31 digits to save the caller from having to manually enter these digits. Dial tone is provided if the system expects to receive more digits from the caller in order to complete the call. If no additional digits are required, the call terminates automatically.

DISA Recorded Announcement (RAN)

A caller may be greeted with a Recorded Announcement once a DISA DN is accessed. The caller can begin dialing anytime during the greeting, in which case the greeting is stopped and the call is processed. If the Recorded Announcement finishes, dial tone is provided if more digits are expected from the caller to complete the call. As with the case of DISA Digit Insertion, the call terminates automatically if no additional digits are required.

Authcode Last Retry Request

For an Authcode Last Request call, if a caller enters an authorization code (Authcode) that is invalid, the caller is prompted to enter an Authcode again. The reprompt for the Authcode takes the form of either an Authcode Last Retry Request dial tone or a RAN before the Authcode Last Retry Request dial tone.

If configured, the RAN indicates to the caller that a wrong Authcode has been entered. While RAN is being given, all dialed digits are ignored.

If a caller realizes they have misdialed, an octothorpe (#) can be pressed which allows the user to immediately re-enter the Authcode. If an invalid Authcode is entered for a second time, the existing invalid Authcode treatment results.

Operating parameters

DISA Digit Insertion, DISA RAN, and Authcode Last Retry can be activated individually or can be combined to work in conjunction with one another.

DISA Digit Insertion and DISA RAN can be optionally assigned on a per DISA basis in LD 24, and are only applicable to DISA calls.

Authcode Last Retry can be optionally assigned on a per customer basis in LD 88, and is applicable to all call types supporting Authcode Last.

All existing DISA limitations apply to the DISA Digit Insertion and DISA RAN functionalities.

All existing RAN limitations apply to the DISA RAN and Authcode Last Retry functionalities.

All existing Authcode Last limitations apply to the Authcode Last Retry functionality.

To support DISA RAN and the Authcode Last Retry RAN function, the system must be equipped with all the necessary RAN hardware.

Feature interactions

Attendant Console Operation

Authcode Last Retry Not Configured

If an invalid Authcode is entered by an attendant, overflow tone is given as soon as a sufficient number of Authcode digits has been entered. If the attendant enters some digits for an Authcode that is less than the number of digits defined in LD 88, silence is heard.

Authcode Last Retry Configured

If the caller is an attendant and the Authcode entered is invalid, once a sufficient number of digits has been entered, the Authcode Last Request dial tone is immediately given to reprompt for the Authcode. If the attendant enters some digits for an Authcode that is less than the number of digits defined in LD 88, silence is heard. Since there is no interdigit time out for an Attendant Console, no Authcode Last Request dial tone will be given for retry.

Authcode Last Request tone will be heard immediately prompting for Authcode Retry if the attendant enters an octothorpe “#” followed by some digits.

Authorization Code Security Enhancement

Only when an Authcode retry fails will a Security Administration (SECA) message be printed to the configured MTC, FIL console and/or the configured History File.

Autodial

If Autodial is programmed with a valid Authcode for Authcode Last followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Call Detail Recording

Digits inserted by DISA Digit Insertion are reflected in the Call Detail Recording (CDR)record.

When a caller is reprompted for an Authcode due to Authcode Last Retry, and a new Authcode is entered, the second Authcode will overwrite the first entry. Therefore, the CDR record only reflects the last Authcode entered.

Pretranslation

Digits automatically inserted by DISA Digit Insertion are pretranslated during call processing in the same manner as if the caller had manually dialed the digits.

Speed Call

If a Speed Call entry is programmed with a valid Authcode for Authcode Last followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Feature packaging

This feature is packaged under Direct Private Network Access (DPNA) package 250.

DISA Digit Insertion requires the following additional package:

- Direct Inward System Access (DISA) package 22

DISA RAN requires the following additional packages:

- Direct Inward System Access (DISA) package 22
- Recorded Announcement (RAN) package 7

Authcode Last Retry requires the following additional packages:

- Basic Authorization Code (BAUT) package 25
- Network Authorization Code (NAUT) package 63
- Recorded Announcement (RAN) package 7 when an Authcode Last Retry RAN is required

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 24 – Modify the Direct Inward System Access data block.

2 LD 88 – Modify the authorization code data block.

DISA DN Data

Configure RAN routes (LD 16) and RAN trunks (LD 14) as per existing procedures.

LD 24 – Modify the Direct Inward System Access data block.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	DIS	DISA data.
CUST	xx	Customer number, as defined in LD 15
...		
RANR	 0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems

- RTMR	(0)-10-300	The maximum amount of time (in seconds) that a caller can wait for an available RAN trunk before being removed from the RAN queue and proceeding as if DISA RAN has been completed.
	(0)	Removes and deactivates the timer.
DGTS	x...x	Digits for DISA Digit Insertion. Up to 31 digits can be defined.
	(X)	Removes and deactivates DISA Digit Insertion.
- DLTN	(YES)	Dial tone needed after digit insertion.
	NO	Dial tone not needed after digit insertion.

Authcode Data

Configure RAN routes (LD 16) and RAN trunks (LD 14) as per existing procedures.

LD 88 – Modify the authorization code data block.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	AUB	Authcode data.
CUST	xx	Customer number, as defined in LD 15
...		
RANR	 0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems
RTRY	(NO)	Disable Authcode Last Retry.
	YES	Enable Authcode Last Retry.

- RAN2	 0-511 0-127 X	Route number for Authcode Last Retry RAN. For Large Systems For Small Systems and Succession 1000 systems Removes and deactivates Authcode Last Retry RAN.
CLAS	xx	Class code value assigned to authcode.

Feature operation

Operational Sequence of a DISA Call

Step	User Action	Result
1.	Dials DISA DN.	If DISA Security Access Code is required, special dial tone is given, and the caller continues to Step 2. Otherwise the caller skips to Step 3.
2.	Enters the Security Access Code.	The dial tone is removed as soon as the first digit is dialed. If the security access code entered is valid, the caller continues to Step 3. Otherwise, the existing treatment for invalid Security Access code is given when the interdigit timer expires.
3.	<no user action>	If Authcode is required, normal dial tone is given, and the caller continues to Step 4. Otherwise, the caller skips to Step 5.
4.	Enters an Authcode.	The dial tone is removed as soon as the first digit is dialed. If the Authcode entered is valid, the caller continues to Step 5. Otherwise, the existing invalid Authcode treatment is given when the interdigit timers times out.
5.	<no user action>	If DISA Digit Insertion is not configured, the caller immediately continues to Step 6. Otherwise, the digits defined for DISA Digit Insertion are automatically inserted into the call register before the caller continues to Step 6.
6.	<no user action>	If DISA RAN is configured, a RAN greeting is provided, and the caller continues to Step 7. Otherwise, the caller skips to Step 8.

7.	a) The caller listens to the RAN greeting; or	a) If DISA Digit Insertion is not defined, or DISA Digit Insertion specifies to give dial tone to prompt the caller to enter more digits, the caller continues to Step 8. Otherwise, the inserted digits are immediately processed for call completion.
	b) begins dialing before the RAN is finished.	b) The RAN greeting is stopped as soon as the first digit is dialed. The dialed digits are appended into the call register (that is, if DISA Digit Insertion is defined, the dialed digits are stored after the inserted digits), and the call is processed for call completion.
8.	<no user action>	Dial tone is given and the caller continues to Step 9.
9.	Dials digits to originate the call.	Dial tone is removed as soon as the first digit is dialed. The dialed digits are appended into the call register (that is, if DISA Digit Insertion is defined, the dialed digits are stored after the inserted digits), and the call is processed for call completion.

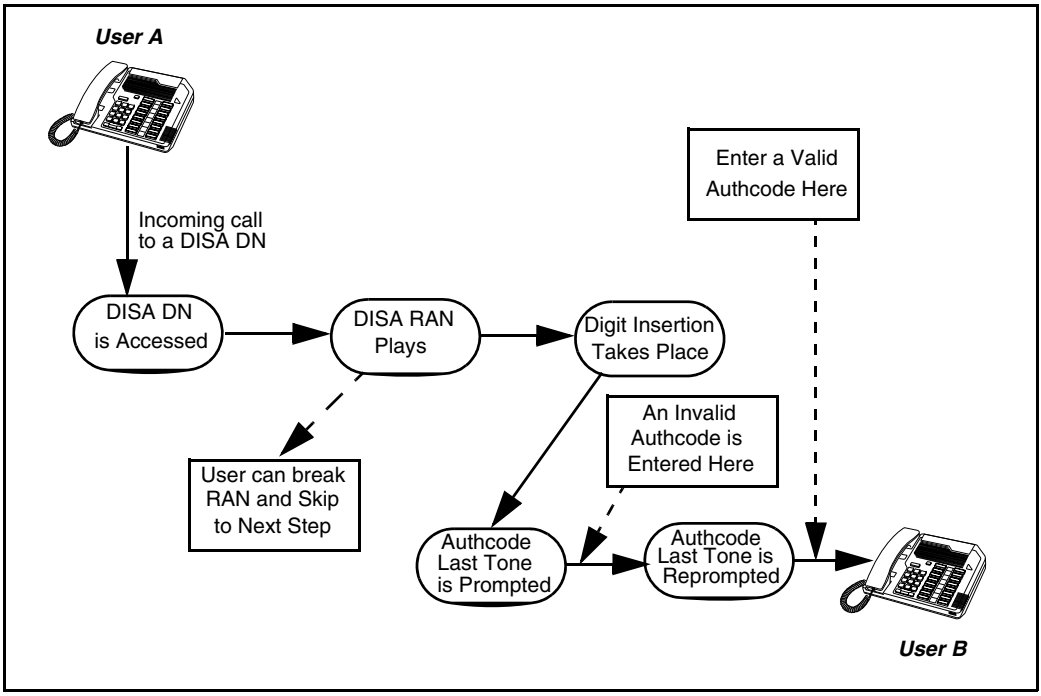
Operational Sequence of Authcode Last

Step	User Action	Result
1.	Makes an outgoing call that requires Authcode Last.	Authcode Last Request dial tone is given. If Authcode Last RAN is defined, RAN precedes the dial tone. The caller continues to Step 2.
2.	Dials one of the following:	The Authcode Last Request dial tone is removed as soon as the first digit is dialed. Then depending on the digit input, one of the following occurs:
	a) A valid Authcode.	a) The call is processed for call termination.
	b) An invalid Authcode followed by "#".	b) If Authcode Last Retry is defined, Authcode Last Request dial tone is immediately given (if Authcode Last Retry RAN is defined RAN precedes the dial tone), and the caller continues to Step 3. If Authcode Last Retry is not defined, when the interdigit timer expires the existing invalid Authcode treatment is given.
	c) An invalid Authcode.	c) If Authcode Last Retry is defined: — If the caller is an attendant, Authcode Last Request dial tone is immediately given (if Authcode Last Retry RAN is defined RAN precedes the dial tone), and the caller continues to Step 3. — If the caller is not an attendant, when the interdigit timer expires Authcode Last Request dial tone is again given (if Authcode Last Retry RAN is defined RAN precedes the dial tone), and the caller continues to Step 3. If Authcode Last Retry is not defined, when the interdigit timer times out the existing invalid Authcode treatment is given.
3.	Dials one of the following:	The Authcode Last Request dial tone is removed as soon as the first digit is dialed. Then depending on the digit input, one of the following occurs:
	a) A valid Authcode	a) The call is processed for call termination.
	b) An invalid Authcode followed by "#".	b) When the interdigit timer times out, the existing invalid Authcode treatment is given.
	c) An invalid Authcode.	c) When the interdigit timer times out, the existing invalid Authcode treatment is given.

Example of a DPNA Call Using All Three Functions

In this example, User A calls from home to a DISA DN and subsequently to an ESN number as defined in the DISA Digit Insertion. When prompted for an Authcode, User A initially enters an invalid one, before being reprompted for the authcode (See Figure 1).

Figure 1
DPNA call using all three functions



Directory Number

Refer to the following feature modules in this book for information on Directory Number:

- "Directory Number Delayed Ringing" on page 177
- "Directory Number Expansion" on page 183
- "Flexible Attendant Directory Number" on page 341
- "Listed Directory Numbers" on page 759
- "Multiple Appearance Directory Number" on page 1081
- "Prime Directory Number" in *Features and Services* (553-3001-306), Book 3
- "Single Appearance Directory Number" in *Features and Services* (553-3001-306), Book 3

For Network-Wide Listed Directory Number, refer to *ISDN Primary Rate Interface: Features* (553-3001-369).

Directory Number Delayed Ringing

Contents

This section contains information on the following topics:

Feature description 177
Operating parameters 178
Feature interactions 178
Feature packaging 181
Feature implementation 181
Feature operation 182

Feature description

There are two types of Directory Number keys: ringing and non-ringing. The Directory Number Delayed Ringing (DNDR) feature offers the ability to provide an audible notification (for example, ringing or buzzing) after a specified delay to non-ringing keys for a particular Terminal Number (TN). These keys can be either Single Call Non-Ringing (SCN) or Multiple Call Non-Ringing (MCN).

When an incoming call is presented to an SCN/MCN key, the associated lamp starts flashing. If Directory Number Delayed Ringing is defined for the set, an audible notification is given after a defined number of seconds (from 1 to 120 seconds). The DNDR value is defined in LD 11, and the feature is disabled if zero is selected as the delay value. When the feature is disabled, all Single Call Non-Ringing (SCN) or Multiple Call Non-Ringing (MCN) keys for this particular TN will not receive audible notifications.

Operating parameters

Only Meridian 1 proprietary telephones with DN key type SCN or MCN may use this feature; analog (500/2500 type) telephones are not supported.

When enabling the Directory Number Delayed Ringing feature and zero is entered as delay value, the desired Single Call Ringing or Multiple Call Ringing key must to changed to Single Call Non-Ringing (SCN) or Multiple Call Non-Ringing (MCN).

The DNDR feature is enabled on a TN basis. Thus, all or none of the SCN/ MCN keys for the TN will receive the audible notification.

For a single call, two appearances of a Multiple Appearance Directory Number (MADN) may ring simultaneously if their DNDR values differ by two seconds or less.

The DNDR value can be different for multiple TNs with the same DN appearance; therefore, the audible notification may begin at different times for a single call.

Feature interactions

Any feature that works with non-ringing keys works with the DNDR feature whether or not the key is ringing.

Attendant Administration
Automatic Wake-Up

Attendant Administration and the Automatic Wake-Up features are not supported.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the Directory Number Delayed Ringing feature and ring the blocked DN immediately when the SACP key is pressed to ring the blocked DN.

Attendant Recall
Automatic Timed Reminder Recalls

If a dialed set has DNDR defined, and an attendant re-extends a call without releasing it, the DNDR timing is not reset. If the value of the recall timer is less than that of the DNDR timer, the call is recalled to the attendant before audible notification begins.

Attendant Recall Enhancement

With this feature, when a call to a set is recalled to the attendant, the ringing is stopped on that set. If the attendant re-extends the call and ringing is applied again, the DNDR delay is also applied.

Buzzing

If a set is defined with DNDR delay and there is an incoming call to another SCN/MCN DN key on the same set, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Call Forward No Answer
Call Forward No Answer, Second Level

The DNDR feature allows the SCN/MCN (non ringing keys) to actually ring after a definable period of time (DNDR prompt in LD 11). If the time before CFNA takes effect is less than the DNDR time for a particular set, CFNA will forward this call before any SCN/MCN keys can ring on this set. Note that CFNA is defined in the number of rings and DNDR is defined in seconds.

If the Forward DN set is busy or invalid when the call is forwarded, the call will return to the originally called set. However, the DNDR delay timer will be reapplied to the called set if DNDR is defined.

If a call is forwarded, as per existing operation, this call will be treated as a new incoming call to the forward DN. For example, if the forward DN has a DNDR value defined, a new timer will begin timing according to the forward DN's DNDR delay.

Call Waiting

Call Waiting tones apply to SCN/MCN keys as per existing operation. The DNDR delay does not apply, and the user is informed of the incoming call immediately.

Data Calls
Private Line Ringing (PVN)
Private Line Non-Ringing
Set-Based Administration Enhancements

These features are not supported by the Directory Number Delayed Ringing feature.

Distinctive/New Distinctive Ringing

The DNDR feature applies to the Distinctive Ringing feature; what applies to normal ringing with DNDR also applies to distinctive ringing.

Flexible Incoming Tones

If DNDR is enabled, the Flexible Incoming Tones buzz is delayed as with any type of audible notification.

Group Call

When a group call is made to an SCN/MCN key with Directory Number Delayed Ringing (DNDR) defined, audible notification will be given after the DNDR delay has expired.

Ringing Change Key

If an SCR/MCR key is toggled from “ringing” to “non-ringing”, the DNDR feature will apply to the key. If an SCR/MCR key is toggled again from “non-ringing” to “ringing”, the key will be rung immediately and DNDR will no longer apply.

If an SCN/MCN key is toggled from “non-ringing” to “ringing”, the DNDR key will ring immediately and DNDR will no longer apply. If an SCN/MCN is toggled again from “ringing” to “non-ringing”, the key will not ring immediately and the DNDR feature will apply to the key.

Short Buzz for Digital Telephones

If a set is defined with DNDR delay and there is an incoming call to another SCN/MCN DN key on the same set, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Spanish KD3 Forced Disconnect

Spanish KD3 Digital Trunk Signaling Direct Inward Dialing (DID) disconnects an incoming call if the destination does not answer in 60 seconds. If the DNDR delay is set to a value of more than 60 seconds, the KD3 DID will terminate the call and the destination never receives the audible notification.

User Selectable Call Redirection

With User Selectable Call Redirection (USCR) a user can change the number of CFNA/DFNA ringing cycles. If the user changes the CFNA/DFNA value so that CFNA takes place before the DNDR timer runs out, none of the SCN/MCN keys will receive an audible notification. See the interaction with Call Forward No Answer.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Configure the delay value (in seconds).

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		

DNDR	(0)-120	Delay value in seconds. A DNDR value of zero disables the feature. If the DNDR value is an odd number, it is rounded up to the next even number. If REQ = NEW, the delay value is 0 (the default); otherwise the existing value appears.
...		
KEY	xx SCN yyyy xx MCN yyyy	Key number, Single Call Non-Ringing, DN. The key must be SCN or MCN.

Feature operation

No specific operating procedures are required to use this feature.

Directory Number Expansion

Contents

This section contains information on the following topics:

Feature description 183
Operating parameters 184
Feature interactions 185
Feature packaging 188
Feature implementation 188
Feature operation 188

Feature description

This feature increases the number of digits allowed for internal Directory Numbers (DNs), from a maximum of four digits per DN to seven digits per DN. The following internal DNs have been expanded:

- Single-line telephone DNs
- Multi-line telephone DNs
- Trunk Group Access codes
- Attendant DNs (including local attendant in Centralized Attendant Service)
- Listed Directory Numbers (LDNs)
- Coordinated Dialing Plan (CDP) steering codes
- Automatic Call Distribution (ACD) DNs

- ACD position IDs
- Direct Inward System Access (DISA) DNs
- Centralized Attendant Service (CAS) hold DNs
- Release Link Trunk (RLT) DNs in Centralized Attendant Service)
- System Park DNs
- Test line DNs, and
- Data service DNs.

The following DN types are not expanded:

- Special Prefix (SPRE)
- Basic/Network Alternate Route Selection (BARS/NARS) access codes
- Route Selection Automatic Number Identification (RSANI) access code, and
- Automatic Modem Pooling (AMP) all-digital-connection prefix.

Along with Directory Number Expansion (DNXP), Call Detail Recording Expansion (CDRE) package 151 is available to allow Call Detail Recording (CDR) records to accommodate the increased digit field lengths. Call Detail Recording (CDR) package 4 and Directory Number Expansion (DNXP) package 150 are required for CDRE.

Operating parameters

The number of DNs that can be configured is limited by the available protected data store in the system.

DNXP does not enhance existing feature capability other than allowing an internal DN with up to seven digits.

If DNXP is equipped, the system communicates with any attached Auxiliary Processor (AUX), except ACD-D, in a new message format containing expanded DN fields. Therefore, the respective Auxiliary Processor (AUX) software must be upgraded to handle longer DNs in new messages.

If a message is sent to an Auxiliary Processor (AUX) that is not capable of handling expanded DNs, only the last four digits are included in the message.

Incoming Digit Conversion (IDC) translates a maximum of four digits only.

The Automatic Number Identification (ANI) calling number is always seven digits long. It is obtained by combining the Automatic Number Identification Listed Directory Number (ANI LDN) with one of the following:

- DN of the analog (500/2500 type) telephone
- Prime DN of the SL-1 telephone
- Automatic Number Identification (ANI) attendant number, specified on a per customer basis, and
- Automatic Number Identification (ANI) trunk number, specified on a per trunk group basis.

With the DNXP package equipped, if an Automatic Number Identification Listed Directory Number (ANI LDN) is not defined, then the full seven digits of an internal DN can be used as the ANI calling number. If an ANI LDN is defined and internal DNs are longer than four digits, only the leading digits of the DNs are retained in the ANI calling number.

CDRE must be equipped to allow the printing of seven-digit DNs in the CDR records. CDRE is not supported by Mini-CDR.

An Automatic Identification of Outward Dialing (AIOD) station identification number remains four digits long. If a DN is longer, only the leading digits are retained as the Automatic Identification of Outward Dialing (AIOD) station identifier.

Service-change and print overlays with DN-related prompts and commands have been modified to accommodate seven-digit DNs if the DNXP package is equipped.

Feature interactions

ACD-C Reports

When the DNXP package is equipped, each DN-related field is expanded to seven digits.

ACD Load Management

ACD Load Management commands have been modified to allow longer DN-related fields (ACD DN, position ID, route access code).

Automatic Identification of Outward Dialing

The Automatic Identification of Outward Dialing (AIOD) station identifier and trunk identifier remains four digits long. If the total number of digits in the AIOD prefix and internal DN exceeds four, only the leading digits of the station DN are retained as the AIOD identifier.

Automatic Number Identification

If the DN Expansion package is equipped, the Automatic Number Identification billing number (ANAT) can have up to seven digits. The total number of digits for ANAT and Automatic Number Identification listed DN (ANLD) cannot exceed seven.

Auxiliary processors

Any AUX or application processor that shares or exchanges system internal DN-related information with the system must be modified to handle the longer DN format. Otherwise, only the four trailing digits will be included in the message.

The presence of DNXP has an impact on the following types of AUX:

- Auxiliary Processor Link (APL)
- Application Module Link (AML)
- Standard Serial Data Interface (SDI) with application interface to the system, and
- Standard SDI without application interface to the system.

Background Terminal Interface

When the DNXP package is equipped, any background terminal command, response, or display containing a DN is allowed to have a DN of up to seven digits.

Coordinated Dialing Plan

Coordinated Dialing Plan (CDP) steering codes are expanded to a maximum of seven digits. The maximum number of digits for a complete CDP DN has increased from seven to ten (a three-digit steering code followed by a seven-digit internal DN).

With DNXP, the maximum number of leading digits to be deleted from a Local Steering Code (LSC) is expanded to seven digits, due to longer CDP numbers.

Digit and Name Display

If longer DNs are defined, the left most digits may be scrolled out on a digit display, depending on the size of the display window.

Direct Inward Dialing

Depending on the number of Direct Inward Dialing (DID) digits outpulsed by the Public Exchange/Central Office (CO), the system can insert a unique string of prefix digits to the incoming Direct Inward Dialing (DID) digits on a per DID trunk group basis to form a final internal DN. The number of digits that can be inserted for a DID (or TIE) trunk group has been expanded from six to eight digits.

Do Not Disturb

If the Directory Number Expansion (DNXP) package is equipped, DNs can have up to seven digits.

Electronic Switched Network

With DNXP, a seven-digit Location Code (LOC) call to an Electronic Switched Network (ESN) switch can be terminated to an internal DN of up to seven digits. A Digit Manipulation Index associated with a Home Location Code is used to properly terminate the calls.

Flexible Attendant Directory Number

The attendant DN can have up to seven digits if the Directory Number Expansion (DNXP) package is equipped.

Integrated Services Digital Network

Refer to *ISDN Primary Rate Interface: Features* (553-3001-369).

Night Service

If the Directory Number Expansion (DNPX) package is equipped, the Night DNs can be up to seven digits; otherwise, the DN can be a maximum of four digits.

Single Appearance Directory Number

The DN can have up to seven digits if the Directory Number Expansion package is equipped.

Feature packaging

Directory Number Expansion (DNXP) package 150 has no other feature package dependencies.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Distinctive/New Distinctive Ringing

Contents

This section contains information on the following topics:

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Operating parameters 190
Feature interactions 190
Feature packaging 192
Feature implementation 193
Feature operation 195

Feature description

In commercial applications, the ability to have telephones with a distinctive ring is useful for distinguishing various call types. The Distinctive Ringing capability is enabled for specific trunk groups.

The Tone and Digit Switch (TDS) card provides Meridian 1 proprietary telephones with distinctive ringing cadence. This card provides a distinctive ringback tone of 440 Hz + 480 Hz on incoming calls on the designated trunks, timed for 1.64 on and 0.36 off. On single-line telephones, the normal ringing pattern is 2 on and 4 off. Distinctive Ringing for single-line telephones is 1.54 on and 0.38 off.

New Distinctive Ringing

This feature provides a new ringing cadence of 0.512 on and 0.512 off, followed by 1.024 on and 4.096 off, for all telephone types.

Distinctive Ringing for Dial Intercom

This feature allows a user to differentiate between an incoming call and a Dial Intercom call. The Dial Intercom ringing has a different cadence than regular Directory Number (DN) ringing or Distinctive Ringing.

Distinctive Ringing for Dial Intercom is assignable on a per-customer basis. The cadence is 0.5 on and 0.5 off, repeatedly.

Operating parameters

Distinctive Ringing requires 2.5 times as much “on” ringing time as routine ringing. The number of simultaneously ringing lines per ringing generator is reduced according to the proportion of incoming calls that receive Distinctive Ringing. For example, if 50 percent of all calls receive Distinctive Ringing, the number of simultaneous ringing lines is reduced from 20 to 14 per ringing generator.

The QPC609D Fast Tone and Digit Switch card, or a later version of this card, is required to implement the New Distinctive Ringing feature.

Feature interactions

Attendant calls

When an incoming trunk call is extended by an attendant, the terminating extension rings distinctively.

Call Forward Busy

Calls modified by Call Forward Busy are not given Distinctive Ringing as they terminate on the Attendant Console.

Call Forward No Answer, Second Level

The ringing cadence for all telephones in a chain of call redirections remains the same as for the original DN called.

Call Waiting Redirection

The existing Distinctive Ringing Call Forward No Answer feature is applied to calls from a Distinctive Ringing enabled trunk. If such an incoming call is receiving Call Waiting treatment on sets with Distinctive Ringing, Call Forward No Answer (CFNA), and the Call Waiting Redirection feature enabled, the DFNA timer is applied to the call instead of the CFNA timer. The Call Waiting warning tone, if enabled, is not changed by Distinctive Ringing. If that call is not answered before the expiration of the DFNA timer, CFNA treatment is given using the Call Waiting Redirection feature.

Directory Number Delayed Ringing

The Directory Number Delayed Ringing (DNDR) feature applies to the Distinctive Ringing feature; what applies to normal ringing with DNDR also applies to distinctive ringing.

Flexible Tones and Cadences

With the Flexible Tones and Cadences package equipped, the SL-1 Call Park Recall Ring Cadence (RBCS) specified in LD 56 has precedence over the Distinctive or New Distinctive Ringing given for Call Park recall.

ISDN Semi Permanent Connections for Australia

For ISDN Semi Permanent Connections for Australia (ISPC) calls, Distinctive/New Distinctive Ringing is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Night Service

Incoming calls terminating on a night Directory Number (DN) ring distinctively.

Telephones

The Meridian digital telephone Distinctive Ringing (defined by the Class of Service in LD 11) specifies the frequency and the warble-tone rate, and does not pertain to the Distinctive Ringing feature as referred to in this feature description.

For example, suppose New Distinctive Ringing is enabled and a call comes in from a Distinctive Ringing enabled trunk. If the call terminates on a Meridian digital telephone with DR2 Class of Service, it rings with DR2 (frequency and warble tone), but with a cadence of 0.512 on and 0.512 off, followed by 1.024 on and 4.096 off. This also applies to the M3000 Touchphone. If the M3000 custom ringing option is selected, Distinctive Ringing is overridden.

Telephone features

Calls modified by the following features receive Distinctive or New Distinctive Ringing:

- Call Forward All Calls
- Call Forward No Answer
- Flexible Call Forward No Answer
- Call Park
- Call Transfer
- Conference
- Hunting

User Selectable Call Redirection

The single parameter previously used to define distinctive ringing cycles (DFNA) is expanded to three (DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific DFNA entry for each telephone.

Virtual Network Services

An incoming call using VNS on a Bearer trunk defined with the prompt DRNG = YES will ignore this value and will perform the treatment as if the value of this prompt was DRNG = NO.

Feature packaging

Distinctive/New Distinctive Ringing (DNRG) package 74 has no feature package dependencies.

Distinctive Ringing for Dial Intercom is included in Dial Intercom (DI) package 21.

Distinctive Ringing for digital telephones is included in Digital Telephones (DSET) package 88.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable or disable Distinctive Ringing for Dial Intercom calls and specify Call Forward No Answer timing for trunks with Distinctive Ringing.

2 LD 17 – Specify Distinctive or New Distinctive Ringing.

3 LD 16 – Enable or disable Distinctive Ringing for each incoming or incoming/outgoing trunk route.

4 LD 11 – Specify Distinctive/New Distinctive Ringing Class of Service for Meridian 1 proprietary telephones.

LD 15 – Enable or disable Distinctive Ringing for Dial Intercom calls and specify Call Forward No Answer timing for trunks with Distinctive Ringing.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- IRNG	(NO) YES	(Disable) enable Distinctive Ringing for Dial Intercom calls.
DFNA	1-(4)-15	The number of distinctive ringing cycles before Call Forward No Answer is activated for calls with Distinctive Ringing (the default is 4).

LD 17 – Specify Distinctive or New Distinctive Ringing.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System parameters.
PARM	(NO) YES	Change system parameters.
- NDRG	(NO) YES	(Disable) enable New Distinctive Ringing (DRNG). Prompted only if DRNG is equipped.

LD 16 – Enable or disable Distinctive Ringing for each incoming or incoming/outgoing trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
DRNG	(NO) YES	(Disable) enable Distinctive Ringing for incoming calls.

LD 11 – Specify Distinctive/New Distinctive Ringing Class of Service for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	DRGX	Distinctive ring type (DRG1), DRG2, DRG3, DRG4, where: DRG1 = high fast tone, frequency 667/500 Hz. DRG2 = high slow tone, frequency 667/500 Hz. DRG3 = low fast tone, frequency 250/333 Hz. DRG4 = low slow tone, frequency 250/333 Hz. The DRG3/4 distinctive ringing for M2006 and M2008 telephones are different: DRG3 = low fast tone, frequency 1600/2000 Hz. DRG4 = low slow tone, frequency 1600/2000 Hz.

Feature operation

No specific operating procedures are required to use this feature.

Distinctive Ringing by DN

Contents

This section contains information on the following topics:

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Operating parameters 198
Feature interactions 199
Feature packaging 201
Feature implementation 202
Feature operation 204

Feature description

Distinctive Ringing by DN (DRDN) allows a distinctive ringing cadence to be configured for each DN key. The ability to have sets with a distinctive ring is useful for distinguishing calls with different DNs and is only available on Meridian Modular sets.

Distinctive ringing is an enhancement to the existing Executive Distinctive Ringing (EDRG) feature. This existing feature supports a distinctive ringing cadence when a call is made from an executive set. The Distinctive Ringing by DN feature enhances the EDRG feature by introducing two new functionalities.

The EDRG feature is determined by Class of Service as executive and it will ring distinctively. The existing functionality of EDRG is modified to allow the ringing cadence to be defined on a DN key basis rather than a TN basis.

A sub prompt for every DN key configures distinctive ringing index for incoming and outgoing calls. There are two available features for incoming and outgoing calls:

- **Distinctive Ringing by call source, per DN-key:** The distinctive ringing given to the called set is determined by the call source (calling set). This functionality is the same as the EDRG feature, except it is DN-key based rather than set based
- **Distinctive Ringing by call destination, per DN-key:** The distinctive ringing given to the called set is determined by the call destination (called set) and is also based on the DN-key of the called set.

With these enhancements, a DN-key can be configured to give a distinctive ring to the terminating set, and receive a distinctive ring for incoming calls.

Operating parameters

The precedence order for the different distinctive ringing cadences to ring the terminating set in a call is:

- Distinctive Ringing for an Incoming trunk call
- Distinctive Ringing by DN by call source
- Executive Ringing by DN call destination
- Distinctive Ringing by DN by call destination

The Private Line Ringing (PVR)/ Non-Ringing (PVN) keys are not supported by the DRDN features.

No DRDN functionality is supported on the Voice Call (VCC) keys since no DN is assigned to a VCC key.

The QPC609D Fast Tone and Digit Switch card, or a later version of this card, is required to implement the New Distinctive Ringing feature.

A total of five distinctive ringing cadences used by DRDN are supported. Therefore a set with more than five DNs will have at least two DN-keys with the same distinctive ringing cadences.

The functionality of DRDN is limited to the following DN-keys; otherwise, normal ringing is given.

- Single Call Ringing (SCR)
- Single Call Non-ringing (SCN)
- One-way HOTLine (HOT)
- Two-way HOTLine
- Conference Hotline (CH)

The following Meridian sets can support DRDN:

- M2006
- M2008
- M2008HF
- M2616
- M2016
- M2216
- M2317
- M3000

Feature interactions

Attendant Extended Call

A call from a set with DRDN extended from the attendant to the called set rings distinctively with the DNRO ringing cadence as configured on the originating set. It the attendant set is not configured for DRDN and the called set is equipped with DRDN then the called set rings with the DNRI ringing cadence as configured on the called set. If DRDN is not configured, normal ringing is given.

Call Forward All Calls

The forwarded call rings distinctively the called set if the originating set is configured with DRDN. If DRDN is not configured on the originating set then the called set rings distinctively, otherwise normal ringing is given.

Call Forward No Answer, Second Level

The ringing cadence for all telephones in a chain of call redirections remains the same as for the original DN called. When CFNA is activated for a set, distinctive ringing is given to the called set if the originator set is configured with DRDN, otherwise normal ringing is given.

Call Transfer

The ringing of the redirected call is determined by the set that has originated the call and not by the set transferring the call. The transferred call distinctively rings the called set if the originating set is configured with DRDN. If the originating set is not configured with DRDN then the ringing of the transferred call is determined by the called set.

Conference

The conference call is either scanned for a call marked as distinctive or a set designed as an executive set. The conferee with the highest index determines the ringing for the new call. The index of the conferees across the network checks if the network supports NAS supplementary messaging.

Dial Intercom Call

A Dial Intercom call is distinguished from a normal call since it has a different cadence configured in the FTC table. Dial Intercom takes precedence over the existing EDRG feature.

Distinctive Ringing

Existing Distinctive Ringing by DN (defined by the Class of Service in LD 11) specifies the frequency and the tone rate where the DRDN features supports the cadences.

Distinctive Ringing by an Incoming Trunk Call

All calling sets marked as distinctive rings the called set with a distinctive ring. The distinctive ring is determined by the index configured for the calling set. This takes precedence over DRDN.

Group Call

Distinctive ringing takes priority over the ringing cadence selected by the DRDN feature.

Hunting

Hunting occurs when the called set is busy. If the originating set is configured with DRDN the called set rings distinctively. A called set on a network call will ring distinctively with the cadence determined by the ringing index received across the network.

Enhanced Hotline

Enhanced Hotline DN-keys are required to support the functionality of the DRDN feature. A call made from Hotline DN-keys rings the called set with the index as configured for DNRO of the key. An incoming call to the HOT key rings the set with the index configured for DNRI.

Flexible Tones and Cadences

With the Flexible Tones and Cadences package 125 equipped, the Call Park Recall Ring Cadence (RBCS) specified in LD 56 has precedence over the Distinctive feature and Distinctive Ringing by DN given for Call Park recall.

Multiple Appearance DN

Distinctive Ringing by DN does not support Multiple Appearance DNs. Therefore, each appearance of a DN configured on a different set cannot be configured to allow different ringing cadences.

Night Service

Incoming calls terminating on a night Directory Number (DN) that has been set up with DRDN ring distinctively. If DRDN is not configured on the calling set, the night DN rings distinctively, otherwise normal ringing is given.

Feature packaging

The following packages are required for Distinctive Ringing by DN:

- Distinctive Ringing (DRNG) package 74

- Flexible Tones and Cadences (FTC) package 125
- Executive Distinctive Ringing (EDRG) package 185

Network Distinctive Ringing (NDRG) for feature functionality over the ISDN requires:

- Distinctive Ringing (DRNG) package 74
- Integrated Service Digital Network (ISDN) package 145
- Integrated Service Digital Network International (ISDN_INTL_SUP) package 161

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 56 – Define the ringing cadence for analog (500/2500 type) sets, and the network and distinctive ringing tone for proprietary sets.

2 LD 11 – Define the distinctive ringing cadence/tone to be used for Meridian 1 proprietary telephones and define Class of Service.

LD 56 – Define the ringing cadence for analog (500/2500 type) sets, and the network and distinctive ringing tone for proprietary sets.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTC	Flexible Tones and Cadence table.
TABL	0-31	FTC table number.
RING	YES	Tones and cadences for ringing.
...	...	
NDR1 PBX	0-(2)-15	Network Distinctive Ring 1 cadence for analog (500/2500 type) sets.

NDR1 BCS		Network Distinctive Ring 1 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- XCAD	0-(2)-15	NT8D17 TDS Cadence code.
NDR2 PBX	0-(2)-15	Network Distinctive Ring 2 cadence for analog (500/2500 type) sets.
NDR2 BCS		Network Distinctive Ring 2 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- XCAD	0-(2)-15	NT8D17 TDS Cadence code.
NDR3 PBX	0-(2)-15	Network Distinctive Ring 3 cadence for analog (500/2500 type) sets.
NDR3 BCS		Network Distinctive Ring 3 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- XCAD	0-(2)-15	NT8D17 TDS Cadence code.
NDR4 PBX	0-(2)-15	Network Distinctive Ring 4 cadence for analog (500/2500 type) sets.
NDR4 BCS		Network Distinctive Ring 4 cadence for proprietary sets.
- XTON	0-(2)-15	NT8D17 TDS Tone code.
- CAD	7	NT8D17 TDS Cadence code.

LD 11 – Define the distinctive ringing cadence/tone to be used for Meridian 1 proprietary telephones and define Class of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type xxxx = 2006, 2008, 2016, 2216, 2317, 2616, 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	d...d	Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	xx	Customer number, as defined in LD 15
...		
CLS	DRDA	Distinctive Ringing by DN enabled. (DRDD) is the default.
...	...	
KEY	xx aaa yyyy	Telephone function key assignments for this feature, where: • xx = key number. • aaa = key type for this feature. These key types include: HOT D (one way and two way hotline), MCR, MCN, SCR, SCN and CH D. • yyyy = Directory Number. ***Note 1:*** The maximum number of distinctive ringing cadences is five. Therefore, a set configured with more than five DNs, say six, can provide distinctive ringing for five of the six DNs. ***Note 2:*** Any call originating from other than the above mentioned keys gives the default ring to the terminating sets.
- MARP	NO	Multiple Appearance DN Redirection Prime.
- DNRO	(0)-4	Distinctive Number Ringing index for outgoing calls.
- DNRI	(0)-4	Distinctive Number Ringing index for incoming calls.

Feature operation

No specific operating procedures are required to use this feature.

Do Not Disturb

Contents

This section contains information on the following topics:

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Operating parameters 208
Feature interactions 208
Feature packaging 212
Feature implementation 212
Feature operation 216

Feature description

Individual Do Not Disturb (DNDI) allows the attendant to place a particular Directory Number (DN) in Do Not Disturb (DND) mode. A DN in this mode is free to originate calls, but appears busy to incoming calls. An attendant dialing a Directory Number in Do Not Disturb mode receives a visual indication and can override it temporarily by using Busy Verify (BVR) and signal source. To activate Individual Do Not Disturb (DNDI), a separate Individual Do Not Disturb (DNDI) key/lamp pair must be assigned to each applicable Attendant Console.

Analog (500/2500 type) telephones can be equipped with a Do Not Disturb lamp. Common Control Switching Arrangement (CCSA) and LPA Class of Service must be allowed.

Calls will receive the customer-specified intercept treatment (for example, busy tone, Recorded Announcement (RAN), or attendant). An enhancement to DND provides the ability to route calls to the Hunt DN instead of to the intercept treatment. Table 2 lists possible intercept treatments based on responses to the prompts Do Not Disturb Intercept Treatment (DNDT) and Do Not Disturb Hunt (DNDH) in LD 15.

Table 2
Do Not Disturb intercept treatments (Part 1 of 2)

Call type	Hunt	DNDT = BST		DNDT = RAN		DNDT = ATT	
		DNDH No	DNDH Yes	DNDH No	DNDH Yes	DNDH No	DNDH Yes
DID							
Analog (500/ 2500 type) telephone	Allow	H	H	R	H	H	H
	Deny	A	A	R	R	A	A
Meridian 1 proprietary telephone	Allow	A	H	R	H	A	H
	Deny	A	A	R	R	A	A
Attendant							
Analog (500/ 2500 type) telephone	Allow	H	H	B	H	H	H
	Deny	B	B	B	B	B	B

H = Follow Hunt Directory Number (DN)
A = Intercept to attendant
B = Busy tone
R = RAN treatment

Table 2
Do Not Disturb intercept treatments (Part 2 of 2)

Call type	Hunt	DNDT = BST		DNDT = RAN		DNDT = ATT	
		DNDH No	DNDH Yes	DNDH No	DNDH Yes	DNDH No	DNDH Yes
Meridian 1 proprietary telephone	Allow	B	H	B	H	B	H
	Deny	B	B	B	B	B	B
Internal							
Analog (500/ 2500 type) telephone	Allow	H	H	R	H	H	H
	Deny	B	B	R	R	A	A
Meridian 1 proprietary telephone	Allow	B	H	R	H	A	H
	Deny	B	B	R	R	A	A

H = Follow Hunt Directory Number (DN)
A = Intercept to attendant
B = Busy tone
R = RAN treatment

Group Do Not Disturb (DNDG) allows an attendant to place predefined groups of DNs in DND mode. A DN can belong to many DND groups.

If a DN belongs to more than one DND group, the DND status of the DN might not be consistent with the DND status of each group. For example, if one of the DN's groups is removed from DND mode, the DN is also removed from DND mode even if another group to which the DN belongs is still in DND mode.

To enable Group Do Not Disturb (DNDG), the DNDI package must be equipped. DNDI allows the user to activate, cancel, and verify the presence of the feature. A separate Group Do Not Disturb (DNDG) key is assigned to each Attendant Console for activating the DNDG feature.

Operating parameters

A maximum of 100 groups (0-99) can be defined per customer. Each group can contain up to 127 DNs.

A maximum of 20 DNDG keys can be equipped on an M2250 Attendant Console. Ten DNDG keys can be equipped on a QCW or M1250 Attendant Console. Alternatively, the DNDI key plus dial-access can be used to activate DND for up to 100 groups.

To activate DNDG using a DNDG key, a group of telephones must be defined for that key (see LD 26).

For Individual Do Not Disturb (DNDI), a Direct Inward Dial (DID) call to a DN with DND active goes to the attendant if DNDT in LD 15 is set to BST or ATT. If the attendant is in Night Service, DID calls go to the night DN, if one is specified.

For Group Do Not Disturb (DNDG), if a DN is busy or has DND active, a DID caller gets a busy tone. If DNDT in LD 15 is set to CDB or RAN, and a DN is busy or has DND active, the DID caller gets RAN and then goes to the attendant.

Feature interactions

Attendant Alternative Answering

A DN in the DND mode is free to originate calls but appears busy to incoming calls. Call Forward All Calls takes precedence over DND indication on Attendant Alternative Answering (AAA) DNs.

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the Do Not Disturb feature. If the dialed DN of the set that has the Do Not Disturb feature active is idle, the DN will be blocked and if the DN is busy, busy tone will be heard.

Attendant Break-In

For a telephone with Do Not Disturb in effect, Break-In is temporarily denied to the attendant. The Break-In lamp uses slow flash to indicate this situation. Using the Break-In key prior to dialing the destination DN circumvents this situation. After the Break-In, the telephone returns to its prior status.

Attendant Break-In to Inquiry Calls

The operation of Do Not Disturb is overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Do Not Disturb that may be applied to the set.

Automatic Wake Up

When a telephone is configured for Do Not Disturb, a wake up call can still be presented.

Call Forward All Calls
Hunting

If activated, Call Forward All Calls, Call Forward, Internal Calls and Hunting take precedence over DND busy indication.

Call Forward/Hunt Override Via Flexible Feature Code

Do Not Disturb is not overridden by the Call Forward/Hunt Override Via FFC feature.

Call Park

Calls can be parked on and by DNs in DND mode. When a telephone in DND mode parks a call, the call will not return to the DND telephone. It recalls to the attendant.

Camp-On, Forced

Telephones with Do Not Disturb enabled cannot be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On.

China – Attendant Monitor

If an attendant attempts to monitor a DN which has Do Not Disturb activated and is idle, idle DN treatment is given.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is not allowed if either of these features is active at the requested party.

Directory Number Expansion

If the Directory Number Expansion (DNXP) package is equipped, DNs can have up to seven digits.

Group Hunt

Do Not Disturb (DND) has priority over Group Hunting. Group Hunting will skip over sets with DND active.

Hunting

If activated, Hunting takes precedence over Do Not Disturb busy indication.

Idle Extension Notification

It is not possible to request for Idle Extension Notification towards an extension that has the Do Not Disturb feature activated.

The Idle Extension Notification feature is not supported on DPNSS networks.

It is not possible to request Idle Extension Notification towards an extension that is Second Degree Busy. Idle Extension Notification is only possible on an extension that is First Degree Busy.

It is not possible to set Idle Extension Notification towards a pilot DN.

Intercept Computer Dial from Directory

This feature can be activated for an extension DN as follows:

- Press an idle Loop key, and press the Do Not Disturb Individual (DND IND) key on the Attendant Console.

- Dial a DN from the ICT.
- Press the DND IND key once more, and terminate the procedure by pressing the Release key on the Attendant Console.

The same approach applies when cancelling Do Not Disturb for a set.

To override Do Not Disturb for an extension DN:

- Press an idle Loop key on the Attendant Console.
- Dial a DN from the Intercept Computer (ICT).

Press the DND IND key on the Attendant Console.

ISDN QSIG/EuroISDN Call Completion

An incoming notification overrides a set with Do Not Disturb (DND) activated. Call Completion requests can be applied to sets with the DND feature activated. However, this request does not advance until the DND feature is deactivated.

Last Number Redial

A Hot Line key cannot be redialed using the Last Number Redial feature.

Make Set Busy and Voice Call Override

Voice calls are not allowed on a set with attendant-activated Do Not Disturb.

Meridian Hospitality Voice Services

Individual Do Not Disturb (DND) allows the attendant to place a Directory Number into DND mode. A DN in this mode is free to originate calls, but appears busy to incoming calls. With MHVS equipped, a new prompt (DNDH) allows callers to be redirected to Meridian Mail for voice mail services. A called telephone must have Hunting Allowed (HTA) class of service, and Hunt to Meridian Mail and DNDH in LD 15 must both be set to YES.

Network Individual Do Not Disturb

An attendant may receive a visual indication of the state of a set belonging to Group Do Not Disturb mode, whether this set is located on the local node or any other network node.

Network Intercom

Hot Type I calls ignore the Do Not Disturb feature. Hot Line calls are presented to the defined target, even when DND is activated.

Night Station

A Night Station DN can be placed in DND mode.

Override

Priority Override

Telephones with DND enabled cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Private Line Service

DND cannot be used on Private Lines.

Feature packaging

Do Not Disturb, Individual (DNDI) package 9 has no feature package dependencies.

Do Not Disturb, Group (DNDG) package 16 requires DNDI package 9.

Do Not Disturb Hunt requires Meridian Hospitality Voice Services (MHVS) package 179.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Specify the treatment received by calls to a number in Do Not Disturb mode.

2 LD 26 – Add or change a Group Do Not Disturb.

3 LD 26 – Merge one or more defined Do Not Disturb groups into another DND group, retaining their status as groups.

4 LD 26 – Print Do Not Disturb group data.

5 LD 12 – Add or change Individual or Group Do Not Disturb keys on an Attendant Console.

6 LD 10 – Enable or disable lamp for analog (500/2500 type) telephones.

LD 15 – Specify the treatment received by calls to a number in Do Not Disturb mode.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- DNDL	(NO) YES	Do Not Disturb lamp for analog (500/2500 type) telephones.
TYPE	INT	Intercept Treatment Option.
- DNDT	(BST) ATT RAN	Busy tone treatment for Do Not Disturb (DND) numbers. Attendant treatment for DND numbers. Recorded announcement for DND numbers.
- - RRT	xxx	Route number for the recorded announcement for calls to a DND number (prompted if DNDT = RAN).
TYPE	RDR	Call Redirection.
- DNDH	(NO) YES	(Disallow) Allow Do Not Disturb Hunt.

LD 26 – Add or change a Group Do Not Disturb.

Prompt	Response	Description
REQ	CHG REM	Change, or remove DN in DND group.
TYPE	DND	Do Not Disturb Group data block.
CUST	xx	Customer number, as defined in LD 15
GPNO	0-99	DND group to be added or changed.
STOR	xxx...x	DN to be added or changed in the DND group; repeat to add other DNs.
RMOV	xxx...x	DN to be removed from a DND group. Prompted if REQ = REM.

LD 26 – Merge one or more defined Do Not Disturb groups into another DND group, retaining their status as groups.

Prompt	Response	Description
REQ	MRG CHG REM OUT	Merge DND groups. Add a DND group from a list of merged DND groups. Remove DND group from a merged group. Remove a DND group that consists of a list of merged DND groups.
TYPE	DND	Do Not Disturb Group data block.
CUST	xx	Customer number, as defined in LD 15
GPNO	0-99	Number of the DND group to be created through merging of other DND groups.
GRP1	G0-G99	Number of the first DND group to be merged (total number of members in all merged DND groups cannot exceed 127). Prompted if REQ = MRG.

GRP2	G0-G99	Number of the second DND group to be merged (total number of members in all merged DND groups cannot exceed 127). Prompted if REQ = MRG.
GRP	G0-G99	Number of the DND group to be merged (total number of members in all merged DND groups cannot exceed 127). Prompted if REQ = MRG.
STOR	G0-G99	Specify the number of the DND group to be added to a list of merged DND groups. Prompted if REQ = CHG.
RMOV	G0-G99	Specify the number of the DND group to be removed from a list of merged DND groups. Prompted if REQ = REM.

LD 26 – Print Do Not Disturb group data.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	DND	Do Not Disturb Group data block.
CUST	xx	Customer number, as defined in LD 15
GPNO	0-99 <CR>	DND group to be printed. Print all DND group data.

LD 12 – Add or change Individual or Group Do Not Disturb keys on an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

KEY	xx DDL	Add an Individual Do Not Disturb key, where: xx = 0-19 for M2250 consoles, and xx = 0-9 for M1250 consoles.
KEY	xx GND 0-99	Add a DND group key, where: xx = 0-19 for M2250 consoles, and xx = 0-9 for M1250 consoles.

LD 10 – Enable or disable lamp for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(LPD) LPA (CCSD) CCSA	(Disable) enable lamp. Controlled Class of Service (denied) allowed.

Feature operation

Individual Do Not Disturb

To activate DNDI using the DNDI key (Attendant Console):

1. Select an idle loop key.
2. Press **DNDI**.
3. Dial the DN of the telephone to place into DND mode.
4. Press **DNDI** again. (Ignore status of indicator.)
5. Press **Rls**.

To deactivate DNDI, follow the same steps.

Group Do Not Disturb

There are two ways to activate DNDG: with the DNDG key or with the DNDI key.

To activate DNDG using the DNDG key (Attendant Console):

1. Press **DNDG**. This key already has a defined group assigned to it. The associated indicator remains steadily lit to indicate that all telephones in that DND group are in DND mode.
2. Press **Rls**.

To deactivate DNDG:

- Press **DNDG**.

To activate DNDG using the DNDI key (Attendant Console):

1. Select an idle loop key.
2. Press **DNDI**.
3. Press the **octothorpe (#)** key.
4. Dial the group number.
5. Press **#** again.
6. Press **DNDI** again.
7. Press **Rls**.

Dual Signaling on Analog Trunks

Contents

This section contains information on the following topics:

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Feature description

A telephone user can select any interexchange carrier for any given call by using a Carrier Access Code (CAC). A CAC comprises an Equal Access identifier and a Carrier Identification Code (CIC). Nortel Networks refers to a call preceded by a CAC as an Equal Access call.

The Dual Signaling on Analog Trunks feature allows Dial Pulse signaling and Digitone signaling to be applied separately to incoming and outgoing calls on one trunk. It reduces the number of Digitone Receiver (DTR) units required on the system since these units are no longer necessary for incoming calls on trunks programmed with the new DPDT Class of Service.

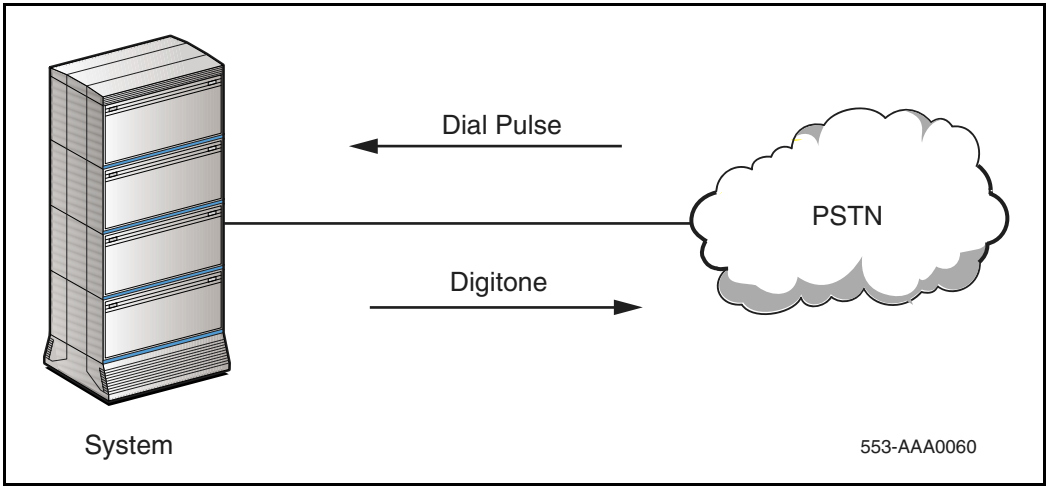
Dual Signaling on Analog Trunks introduces the following trunk Classes of Service in LD 14:

- DPDT = digit information is received as Dial Pulse and sent as Digitone
- DTDP = digit information is received as Digitone and sent as Dial Pulse

Prior to the introduction of Dual Signaling on Analog Trunks, a similar functionality was available when trunks were programmed for DTMF signaling. Dial Pulse calls, if received, were analyzed and handled by the Tone and Digit Switch or Extended Conference and Tone Service card. A DTR was reserved, needlessly, for the duration of the signaling.

The following diagram shows one application of the feature.

Figure 2
System connected to the CO through analog trunks interface



This feature enables a trunk to be configured in one of the following ways:

- incoming Dial Pulse - outgoing Dial Pulse
- incoming DTMF - outgoing DTMF
- incoming Dial Pulse - outgoing DTMF
- incoming DTMF - outgoing Dial Pulse

Operating parameters

The new Classes of Service (DPDT and DTDP) are mutually exclusive with DIP, DTN, MFC, MFE, MFK, MFR and MFX.

If Dual Signaling on Analog Trunks is used on a trunk with DPDT programmed, a DTR is not involved with incoming trunk traffic.

This feature is available on analog DID and TIE trunks only.

CLS DPDT/DTDP can only be configured on routes with the ICOG prompt set to IAO (incoming and outgoing).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 14 – Configure the trunk with the Dual Signaling on Analog Trunks Class of Service.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	aaa	Trunk type. xxx = DID, TIE.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(DIP)	Dial Pulse.
	DPDT	Incoming Dial Pulse - outgoing Digitone.
	DTDP	Incoming Digitone - outgoing Dial Pulse.

Feature operation

No specific operating procedures are required to use this feature.

Electronic Brandlining

Contents

This section contains information on the following topics:

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Feature description

The Electronic Brandlining (EBLN) feature enhances the display functionality of Meridian Modular sets. This feature allows the second line on the idle[1] display screen of a Meridian Modular set to show a custom display.

The display screen of a Meridian Modular set contains two lines with 24 character spaces on each line. Previously, the second line on the display screen of an idle Meridian Modular set was blank. With the Electronic Brandlining feature, however, a custom display is shown left justified on the second line of the idle display screen.

1.Previous to the Electronic Brandlining feature, when a Meridian Modular set is in the idle state, only the time and date is shown on the first line of the display screen and the second line is blank.

Incremental Software Management

An Incremental Software Management (ISM) parameter is introduced with the Electronic Brandlining feature. This ISM parameter is used to transfer custom display information from the Order Management System to system software. The Electronic Brandlining ISM value is copied from the appropriate tape/keycode/file into system software during sysload. The system software then sends the custom display to the display screen of a Meridian Modular set.

The Electronic Brandlining ISM value contains one of the following:

- a Terminal Text Broadcast customized text string value
- a default value

The value of the Electronic Brandlining ISM parameter determines the content of the Electronic Brandlining custom display.

LD 22 is modified to print the Electronic Brandlining ISM parameters. When REQ = SLT (Print System Limits: Incremental Software Management) in LD 22, the ISM parameters, system limits, and keywords are printed. The printing of the Electronic Brandlining custom display output is added after the ISM parameters.

Customers can deliver ISM parameters through keycode. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects ISM parameters.

To expand ISM limits, customers must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in LD 143. To make the expansion effective, the customer must sysload. For further information on keycode installation, please refer to *Large System: Upgrade Procedures* (553-3021-258).

For further information on ISM, refer to the “Incremental Software Management” on page 573 in this book.

Custom Displays

The Electronic Brandlining feature provides the following two custom displays:

- Terminal Text Broadcast Customized Text
- Default “NORTEL” or blank display

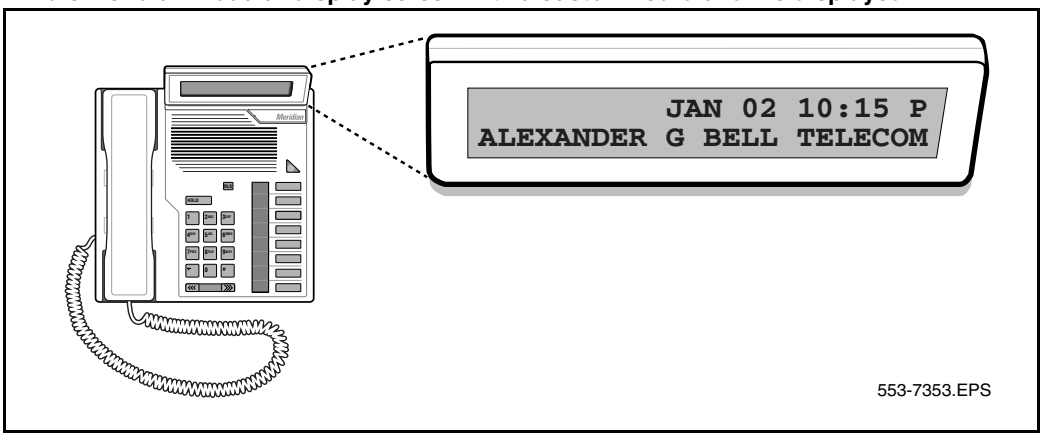
Terminal Text Broadcast Customized Text

When the Electronic Brandlining ISM parameter is equal to the Terminal Text Broadcast value, the customized brandline to be displayed is initially defaulted to NORTEL. This brandline can then be configured to display a different customized brandline.

The customized brandline can have a maximum of 24 characters, each of which must be supported by the North American Meridian Modular set display firmware. Version 18 firmware supports 7-bit ASCII Roman characters and 8-bit non-ASCII Roman characters (See Tables 3 and 4). Alphanumeric and punctuation characters are supported.The customized brandline is configured on a system basis (LD 17).

Figure 3 shows an example of a customized brandline displayed on the idle screen of a Meridian Modular set.

Figure 3
An idle Meridian Modular display screen with a customized brandline displayed



In addition to displaying a customized brandline, the Terminal Text Broadcast functionality can also be used to broadcast a customized text string on the idle display screen of a Meridian Modular set. The text string can have a maximum of 24 supported characters (See Tables 3 and 4). The customized text string is configured on a system basis (LD 17).

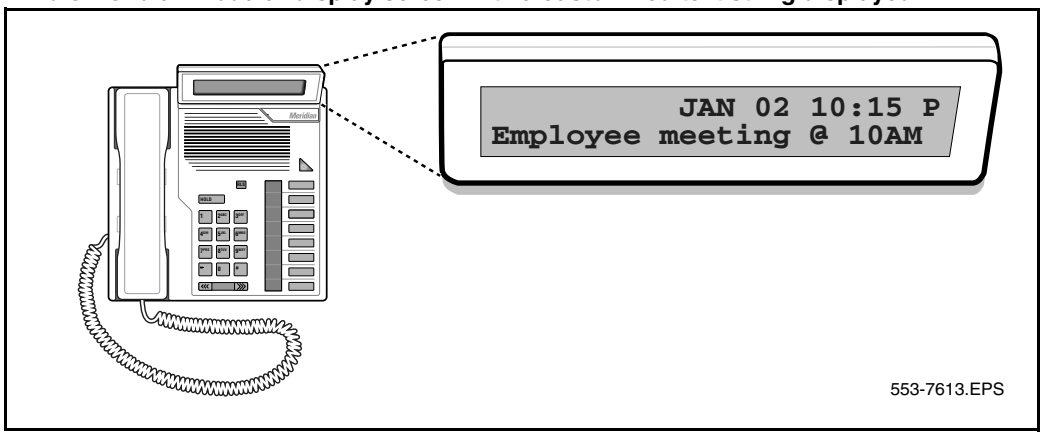
The customized text string can be composed of a single blank space. In this case, the second line of the idle display screen is blank, as per existing functionality.

To enter the customized brandline or text string in LD 17, use one of the following methods:

- Enter a line of supported characters followed by a Carriage Return (<CR>) at the IDLE_DISP_STRING prompt in LD 17.
- Enter a valid character one at a time using either a supported character or its two digit hexadecimal representation at the IDLE_DISP_CHAR prompt in LD 17. The end of input is indicated when only a <CR> is entered or when the 24th character is entered.

Figure 4 shows an example of a customized text string displayed on the idle screen of a Meridian Modular set.

Figure 4
An idle Meridian Modular display screen with a customized text string displayed



Supported characters

Table 3 lists the 7-bit ASCII Roman characters and the corresponding hexadecimal representations that are supported by the Electronic Brandlining feature.

Table 3
Valid 7-bit ASCII Roman Characters

20 <space>	21 !	22 "	23 #	24 $	25 %
26 &	27 ’	28 (	29)	2A *	2B +
2C ,	2D -	2E .	2F /	30 0	31 1
32 2	33 3	34 4	35 5	36 6	37 7
38 8	39 9	3A :	3B ;	3C <	3D =
3E >	3F ?	40 @	41 A	42 B	43 C
44 D	45 E	46 F	47 G	48 H	49 I
4A J	4B K	4C L	4D M	4E N	4F O
50 P	51 Q	52 R	53 S	54 T	55 U
56 V	57 W	58 X	59 Y	5A Z	5B [
5C \	5D]	5E ^	5F _	60 `	61 a
62 b	63 c	64 d	65 e	66 f	67 g
68 h	69 i	6A j	6B k	6C l	6D m
6E n	6F o	70 p	71 q	72 r	73 s
74 t	75 u	76 v	77 w	78 x	79 y
7A z	7B {	7C |	7D }	7E ~	7F ¿

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Table 4 lists the 8-bit non-ASCII Roman characters and the corresponding hexadecimal representations that are supported by the Electronic Brandlining feature.

Table 4
Valid 8-bit non-ASCII Roman Characters

A0 <NASP>	A1 Ľ	A2 ¢	A3 £	A4 Ń	A5 ¥
A6 Ś	A7 Ď	A8 ¨	A9 ©	AA Ň	AB Ť
AC Ź	AD Ÿ	AE ®	AF Ż	B0 °	B1 ±
B2 Ą	B3 Ł	B4 Ř	B5 Ľ	B6 Ć	B7 Č
B8 Ě	B9 Š	BA º	BB Ę	BC Ŕ	BD Ů
BE Ž	BF ¿	C0 À	C1 Á	C2 Â	C3 Ã
C4 Ä	C5 Å	C6 Æ	C7 Ç	C8 È	C9 É
CA Ê	CB Ë	CC Ì	CD Í	CE Î	CF Ï
D0 Đ	D1 Ñ	D2 Ò	D3 Ó	D4 Ô	D5 Õ
D6 Ö	D7 ×	D8 Ø	D9 Ù	DA Ú	DB Û
DC Ü	DD Ý	DE Þ	DF ß	E0 à	E1 á
E2 â	E3 ã	E4 ä	E5 å	E6 æ	E7 ç
E8 è	E9 é	EA ê	EB ë	EC ì	ED í
EE î	EF ï	F0 ò	F1 ñ	F2 ò	F3 ó
F4 ô	F5 õ	F6 ö	F7 ÷	F8 ø	F9 ù
FA ú	FB û	FC ü	FD ý	FE þ	FF ÿ

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Note: Characters that are listed in Tables 3 and 4 are available with North American Version 18 firmware. Individual TTYs may not match the characters and hexadecimal representations in the same way as shown in Tables 3 and 4.

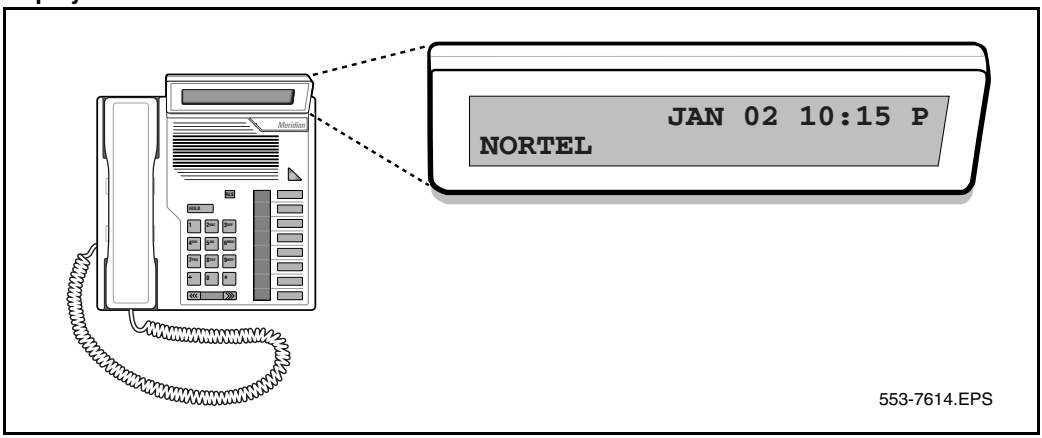
Default Electronic Brandlining Display

If the Terminal Text Broadcast custom display is not chosen, then the Electronic Brandlining ISM parameter value indexes into a default brandline. The default brandline is “NORTEL”, the Meridian Modular set manufacturer (Nortel Networks). This default brandline is displayed left justified on the second line of the idle display screen of a Meridian Modular set.

The default brandline is enabled/disabled on a system basis (LD 17). When the NORTEL_BRAND prompt is set to NO, the second line of the idle Meridian Modular display screen is blank, as per existing functionality.

Figure 5 shows the default brandline (NORTEL) displayed on an idle Meridian Modular set.

Figure 5
An idle Meridian Modular display screen with the Default Electronic Brandlining, “NORTEL” displayed



Operating parameters

The Electronic Brandlining feature applies to Meridian Modular sets that are equipped with a display screen and the appropriate Meridian Modular display firmware. Meridian Modular sets include: M2008, M2016, M2616, M2216ACD1, and M2216ACD2.

The Meridian Modular display firmware, North American Version 18 (Three Language Display) or later, is required for Meridian Modular sets to use the Electronic Brandlining feature. North American Version 18 firmware supports English, French, and Spanish.

The North American Version 18 firmware stores and displays the custom display. If the custom display is sent to a Meridian Modular set without the new firmware, the extra Scan and Signal Distributor (SSD) messages are ignored.

There is an incremental impact of sending SSD messages for a customized brandline. Therefore, it is recommended that no brandlining be done for heavily loaded systems experiencing delays on the High Speed Link (HSL). Instead, the default EBLN brandline can be chosen. Only one SSD message is sent whether the NORTEL_BRAND prompt is set to YES or NO. To minimize the number of SSD messages with the Terminal Text Broadcast custom display, a blank display can be configured. In this case, the customized text string is composed of a single blank space, and only one SSD message is sent for the same real time impact as the default EBLN custom display.

The custom display can have a maximum of 24 characters. Each character must be supported by North American Version 18 firmware.

Version 18 firmware supports 7-bit ASCII Roman characters and 8-bit non-ASCII Roman characters, regardless of whether or not the Multi-language TTY Input/Output (MLIO) package 211 is equipped. Alphanumeric and punctuation characters are supported.

When the MLIO package is restricted, if the “Valid 8-bit non-ASCII Roman Characters” that are supported are used in a custom display, then a 7-bit TTY may not be able to print the characters. If not, then each character is replaced with an underscore character.

If the MLIO package is not restricted and a 7-bit TTY is used, the 8-bit supported characters cannot be printed correctly. Instead, the service change administration interfaces may print garbage characters and/or the interfaces may lock.

When the MLIO package is not restricted, the system sends the valid 8-bit characters to the TTY, rather than the underscore characters. With the MLIO package equipped, it is assumed that the TTY is capable of handling 8-bit characters. If the TTY is capable of entering the "¿" 7-bit character and all other supported 8-bit characters directly, then these characters are accepted by the system, without using the hexadecimal values for the Terminal Text Broadcast customized text. The hexadecimal values can, however, still be used for entries.

The "!" character cannot be entered directly from the TTY keyboard. It can be entered, however, through character-by-character input (IDLE_DISP_CHAR nn prompt in LD 17), using its hexadecimal value.

When the system does not recognize a temporary power outage on a Meridian Modular set, the screen may remain blank until the custom display information, along with the time and date information, is downloaded again.

If the new Electronic Brandlining ISM parameter has an invalid value, the default display is shown. In this case, conversion should have defaulted the NORTEL_BRAND to YES, and as long as this prompt has not been changed, "NORTEL" is displayed.

If the Electronic Brandlining ISM parameter is set to the Terminal Text Broadcast value and the customized text string is configured as "NORTEL" or blank, the NORTEL_BRAND option does not apply. The NORTEL_BRAND option only applies to toggles between "NORTEL" and a blank second line if the Electronic Brandlining ISM parameter is set to an Electronic Brandlining ISM default value.

For new systems, the NORTEL_BRAND prompt is automatically set to YES (default), and the "NORTEL" default brandline is displayed. For the Terminal Text Broadcast option, the NORTEL_BRAND field is automatically set to YES (default); although, the NORTEL_BRAND field is not applicable nor is it output in LDs 17 and 22. The Terminal Text Broadcast customized brandline is initially set to the default "NORTEL" brandline.

No changes are made to the features which currently output information on the second line of the idle display screen of a Meridian Modular set. These features and their output have precedence over the Electronic Brandlining feature. The following idle screens take precedence over the Electronic Brandlining feature: Automatic Answerback, Call Forward, Logged Out, Make Set Busy, Not Ready, and Overflow Busy.

Feature interactions

Automatic Answerback

When Automatic Answerback (AAB) is activated on a Meridian Modular set, the second line of the idle display screen shows “AUTO ANSWER ACTIVATED”.

The Electronic Brandlining custom display is not shown when AAB is activated.

Call Forward All Calls
Internal Call Forward

When Call Forward All Calls or Internal Call Forward is activated on a Meridian Modular set, the second line of the display screen shows “CFWD” on the idle screen. The Electronic Brandlining custom display is not shown when Call Forward All Calls or Internal Call Forward is activated.

When Call Forward All Calls or Internal Call Forward is de-activated on a Meridian Modular set, the second line of the display screen shows “CALL FORWARD CANCELLED” on the idle screen for a few seconds. The Electronic Brandlining custom display is not shown while “CALL FORWARD CANCELLED” is displayed. When the “CALL FORWARD CANCELLED” display times out, the Electronic Brandlining custom display is shown.

Digital Set Display Download

With the Electronic Brandlining feature, the existing time and date messages are modified to include the Electronic Brandlining custom display as part of its data (if applicable).

Display key

When the Display (DSP) key is first pressed, the display screen is blank. When any other key is pressed after the DSP key is pressed, all relevant information is displayed.

The Electronic Brandlining custom display is not displayed during the DSP key process until Lamp Audit updates the display screen with the time and date (when applicable).

Do Not Disturb

When a set is in the Do Not Disturb (DND) mode, the second line of the idle display screen is blank. Therefore, the second line displays the Electronic Brandlining custom display when the Electronic Brandlining feature is enabled (if applicable).

Limited Access to Overlays

The existing functionality of the Limited Access to Overlays (LAPW) feature is not changed as a result of the Electronic Brandlining feature.

The Terminal Text Broadcast configuration of a customized text string in LD 17 is password protected by level 2 system administration (PWD2). The added implementation of PWD2 in LD 17 is required to allow configuration of the Terminal Text Broadcast customized text string.

As per existing functionality, when LAPW is disabled on a system, the PWD2 password is restricted to a 4-digit password composed of the hexadecimal digits 0-9 and/or A-F.

As per existing functionality, when LAPW is enabled, PWD2 can be configured as a 16-digit alphanumeric password. LAPW then applies to the PWD2 prompt.

Make Set Busy

When Make Set Busy (MSB) is activated on a Meridian Modular set, the second line of the idle display screen shows “SET BUSY ACTIVATED”.

The Electronic Brandlining custom display is not shown when Make Set Busy is activated.

Set Based Administration

When a service change is made by Set Based Administration (SBA), the downloading of the time, date, and the Electronic Brandlining custom display (if applicable) is induced.

Set Relocation

Automatic Set Relocation (ASR) and Modular Telephone Relocation (MTR) include the “plugging in” of a Meridian Modular set for its feature operation. When a Meridian Modular set is “plugged in”, the power-on-reset induces the downloading of the time, date, and Electronic Brandlining custom display (if applicable).

System Access Enhancements

The existing functionality of the System Access Enhancements (SAE) feature is not changed as a result of the Electronic Brandlining feature.

The SAE feature applies to the added implementation of the PWD2 prompt in LD 17 for the Terminal Text Broadcast configuration of a customized text string.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure the NORTEL Electronic Brandline.

2 LD 17 – Enter a customized text string.

3 LD 11 – Enable the display on a Meridian Modular set.

LD 17 – Configure the NORTEL Electronic Brandline.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System Parameters.
...		
NORTEL_BRAND	(YES)	"NORTEL" Electronic Brandline is displayed (default).
	NO	"NORTEL" Electronic Brandline is not displayed.
		NORTEL_BRAND is only prompted when the ISM parameter is set to the default value.

LD 17 – Enter a customized text string.

Note: To enter a customized text string, the Electronic Brandlining ISM parameter must be set to Terminal Text Broadcast.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	System Parameters.
...		
IDLE_SET_DISPLAY aaaa		The current customized text string "aaaa" is shown. This information is displayed for confirmation only.
- - MODIFY		Gateway to new EBLN Terminal Text Broadcast configurations.
	(NO)	Enter NO to keep existing configuration (default).
	YES	Enter YES to prompt for further EBLN Terminal Text Broadcast configuration.

- - PWD2	x...x	Password 2. The second level administration password is needed to allow configuration of the Terminal Text Broadcast customized text string.
- - SUPPORTED_TEXT_ONLY		Change customized text string by text string input.
	(YES)	Enter YES to input by text string, and the IDLE_DISP_STRING prompt is prompted.
	NO	Enter NO to input character by character, and the IDLE_DISP_CHAR nn prompt is prompted.
- - - IDLE_DISP_STRING	bbbb	Enter the customized text string. IDLE_DISP_STRING is prompted only if SUPPPORTED_TEXT_ONLY = YES. A maximum of 24 supported characters are accepted and validated. For a blank display, enter <CR> only.
IDLE_SET_DISPLAY bbbb		The customized text (bbbb) entered at the IDLE_DISP_STRING prompt is shown. This information is displayed for confirmation only. It is confirmed at the following OK prompt.
- - OK		Confirm the validated Terminal Text Broadcast customized text string (bbbb) entered at the IDLE_DISP_STRING prompt.
	(YES)	Enter YES to keep the new text string as “bbbb”.
	NO	Enter NO to input a new Terminal Text Broadcast customized text string, and the Supported_TEXT_ONLY prompt is re-prompted.
...		

- - - IDLE_DISP_CHAR nn		Enter the customized text string character by character.
	c	c = one supported character
	hh	hh = 2 hexadecimal digits (0-9, A-F, a-f), representing a supported character. nn (01-24) is the position of the character in the customized text string. The IDLE_DISP_CHAR prompt is only prompted if SUPPORTED_TEXT_ONLY = NO. It is reprompted until <CR> only is entered or until nn is the 24th character that has been entered.
IDLE_SET_DISPLAY cccc		The customized text string (cccc) entered at the IDLE_DISP_CHAR prompt is shown. This information is displayed for confirmation only. It is confirmed at the following OK prompt.
- - OK		Confirm the validated Terminal Text Broadcast customized text string (cccc) entered at the IDLE_DISP_CHAR nn prompts.
	(YES)	Enter YES to keep the new text string as “cccc”.
	NO	Enter NO to input a new Terminal Text Broadcast customized text string, and the SUPPORTED_TEXT_ONLY prompt is re-prompted.

LD 11 – Enable the display on a Meridian Modular set.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Telephone type, where aaaa is: 2008, 2016, 2216, 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	 (ADD) DDS	Digit Display options Automatic Digit Display (default). Delay Display. When CLS = DDS, the display is activated after the call is answered.
...		

Feature operation

No specific operating procedures are required to use this feature.

Electronic Switched Network

Contents

This section contains information on the following topics:

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Feature description

The Electronic Switched Network (ESN) group of features is designed to support voice and circuit-switched voiceband data telecommunications needs for multiple-location customer applications.

Basic Authorization Code

The Basic Authorization Code (BAUT) feature provides up to 5000 authorization codes of 1 to 14 digits that allow selected users to temporarily override system access restrictions by dialing a Special Service Prefix (SPRE) code, the digit 6, and the Basic Authorization Code (BAUT). The Basic Authorization Code (BAUT) is used for general applications and is described in *Basic Network Features* (553-3001-379).

Basic Alternate Route Selection

Basic Alternate Route Selection (BARS) enables calls placed to another location to be routed automatically over the least expensive route. After the Basic Alternate Route Selection (BARS) access code and the desired number have been dialed, Basic Alternate Route Selection (BARS) automatically tries alternate routes to the destination and completes the call over the least expensive route available at the time of dialing. BARS is described in detail in the *Basic Network Features* (553-3001-379).

Call Back Queuing

Call Back Queuing (CBQ) is an optional feature available to systems equipped with the Basic/Network Alternate Route Selection (BARS/NARS) or Coordinated Dialing Plan (CDP) features. If all facilities are busy when an individual places a BARS, NARS, or CDP call, Call Back Queuing (CBQ) enables the individual to invoke the Ring Again (RGA) feature and receive a callback from the system when a facility becomes available. Call Back Queuing (CBQ) is described in detail in Network Queuing description or *Basic Network Features* (553-3001-379).

Call Back Queuing to Conventional Mains

Call Back Queuing to Conventional Mains (CBQCM) enables call originators at a Conventional Main (any type of switch, including switches that are part of an Electronic TIE Network [ETN]) to access the CBQ feature at the serving ESN Node. When offered CBQ by the Node, users at the Conventional Main dial their extension number to accept the CBQ offer. When facilities become available at the Node, it initiates a CBQ callback to the call originator at the Conventional Main. Refer to *Basic Network Features* (553-3001-379) for a detailed description of Call Back Queuing to Conventional Mains (CBQCM).

Coordinated Call Back Queuing

Coordinated Call Back Queuing (CCBQ) enables telephones eligible for Ring Again (RGA) at the Main to be offered CBQ when network calls are blocked at the serving Node. When facilities become available at the Node, the call originator at the Main is alerted by a callback (identical to an RGA callback) from the Node. Coordinated Call Back Queuing (CCBQ) requires that the Main and associated Node be equipped with Network Signaling. Refer to *Basic Network Features* (553-3001-379) for a detailed description of Coordinated Call Back Queuing (CCBQ).

Coordinated Call Back Queuing Against Main

Coordinated Call Back Queuing Against Main (CCBQAM) is an enhancement to the CCBQ feature that allows a station at the Node to be offered CBQ if a call is blocked at the Main. When facilities become available at the Main, the call originator at the Node is alerted by a callback from the Main. The Network Signaling feature must be equipped at both the Main and the Node for Coordinated Call Back Queuing Against Main (CCBQAM) implementation.

Coordinated Dialing Plan

Coordinated Dialing Plan (CDP) enables a customer with a number of switches to coordinate the dialing plan of stations at these switches. The Coordinated Dialing Plan (CDP) feature allows the telephone user to call any other telephone within a CDP group by dialing a three- to seven-digit number assigned to the station. CDP can be arranged to provide a centralized public exchange network capability that channels access to the public network through a single system switch within the CDP group.

CDP routes Direct Inward Dialed (DID) calls over Central Office (CO) and Wide Area Telephone Service (WATS) trunks using a Distant Steering Code (DSC). The feature is controlled by the Customer Data Block (LD 15). This applies to CO, WATS, Data Terminal Interface (DTI), and Integrated Services Digital Network (ISDN) trunks.

CDP is described in detail in the *Dialing Plans: Description* (553-3001-183).

Flexible ESN “0” Routing

Flexible ESN “0” Routing allows the routing of calls on different routes based on a few predefined non-leftwise unique dialing sequences. “Leftwise unique” means that each entry cannot match the left most portion of any other entry in the table. For example, if “123” is an entry in the table, then no other entry may begin with “123.”

The ESN translation table will allow any or all of the following non-leftwise unique numbers (along with their associated route lists) to be entered into the ESN translation table:

- 0
- 00
- 01
- 011

Flexible ESN “0” Routing is part of the existing BARS (57) and Network Alternate Route Selection (NARS) (58) packages and has no interaction with other features besides these. Since NARS has two translation tables, two Flexible ESN “0” Routing data blocks will be included in NARS. This means that a call could be configured to route in two different ways.

This feature is applicable to all route types and network types supported by ESN. For information on the appropriate prompts and responses in Service Change (LD 90), refer to the *Software Input/Output: Administration* (553-3001-311).

Network Alternate Route Selection

Network Alternate Route Selection (NARS) is an integral part of Nortel Networks’s ESN. Network Alternate Route Selection (NARS) is designed for large business customers with numerous distributed operating locations, enabling the customer to tie together the switches at the various operating locations to create a private telecommunications network. NARS is described in detail in the *Basic Network Features* (553-3001-379).

BARS/NARS Incoming Trunk Group Exclusion

Incoming Trunk Group Exclusion (ITGE) is an enhancement to the BARS/NARS feature. Standard call blocking is applied on outgoing calls to a specific Numbering Plan Area (NPA), NXX, Special Number (SPN), or Location Code (LOC) at the ESN node if the call is from a specific incoming trunk group.

This prevents loopback routing through the caller's home switch (home NPA, NXX). Calls that should have been made off-net from the caller's home switch are blocked outgoing at the Node. Main users are prevented from using the ESN to make calls to certain NPA, NXX, SPN, or Location Codes (LOC) that they are restricted from making at the home switch.

Incoming Trunk Group Exclusion (ITGE) provides full ten-digit restriction for NPA and SPN codes, seven-digit restriction for NXX codes, and three-digit restriction for Location Code (LOC) codes.

Detailed information on this enhancement is provided in *Basic Network Features* (553-3001-379).

NARS Multiple DID Office Code Screening

Multiple DID Office Code Screening is an enhancement to the On-Net to Off-Net Overflow capability of the NARS feature. This enhancement permits network calls that undergo on-net to off-net conversion to terminate at any Directory Number (DN) that has been defined in the LOC data block of memory. This data block allows the definition of multiple office codes (NXX) and/or multiple Directory Number (DN) ranges of the following types:

- single office code/single Directory Number (DN) range
- single office code/multiple DN ranges
- multiple office codes/single DN range
- multiple office codes/multiple DN ranges

NARS Multiple DID Office Code Screening operates within the following parameters:

- Only one Numbering Plan Area (NPA) per LOC is allowed.

- Ranges defined within a LOC must be unique. Overlapping or duplication of ranges is not permitted.
- The number of digits must be the same in each Direct Inward Dialing (DID) range.
- A maximum of 20 Direct Inward Dialing (DID) ranges may be defined per location code.

BARS/NARS Off-Net Number Recognition

Off-Net Number Recognition is an enhancement to the Basic/Network Alternate Route Selection (BARS/NARS) feature for ESN, and for the BARS feature for standalone applications.

Off-Net numbers that terminate at an ESN Node or Main, or at a Conventional Main, can be routed through the private network by means of TIE trunks. BARS/NARS Off-Net Number Recognition prevents unnecessary TO and FROM terminations through CO trunks, at the terminating end, when a caller dials a DID or Direct Distance Dialing (DDD) call to a location in the private network. Calls are handled on the basis of customer-defined parameters stored in Network Translation Tables and Supplementary Digit Recognition/Restriction Blocks.

Detailed information is provided in *Basic Network Features* (553-3001-379).

BARS/NARS 11-Digit Translation

This feature expands the ESN BARS/NARS translation capabilities from a maximum of four digits to a maximum of 11 digits for route selection.

Possible conflicts between translatable codes (NPA, NXX, LOC, SPN) are eliminated by 11-Digit Translation. By allowing translation of more than four leading digits, unique nonconflicting routing to a destination is possible. More than one route list can exist for each specific code of a type. For example, the NXX 727 could only translate into one route list previously.

With 11-Digit Translation, up to 128 route lists for BARS and up to 256 for NARS may be defined, extending translation deeper into the dialed code. The codes must be leftwise unique. If an NXX of 7271 is defined, any other 727 entries must be extended to four digits.

BARS/NARS 11-Digit Translation is discussed in greater detail in *Basic Network Features* (553-3001-379).

Network Authorization Code

The Network Authorization Code (NAUT) feature provides up to 50,000 authorization codes. Network Authorization Code (NAUT) incorporates all the features of the BAUT feature, adds a conditionally last option for entering an Authorization Code after dialing an ESN call, and enables the attendant to enter an Authorization Code. Network Authorization Code (NAUT) is described in detail in *Basic Network Features* (553-3001-379).

Network Call Transfer

Network Call Transfer (NXFER) enhances the operation of Call Transfer (XFER) between two switches when a call is transferred back to the originating switch. The regular Call Transfer feature requires two TIE trunks to complete the call. With Network Call Transfer (NXFER), if the call is transferred back to the originating switch as the same TIE trunk group, the originating switch completes the transfer within itself and the TIE trunks are dropped. For a detailed description of Network Call Transfer (NXFER) refer to *Basic Network Features* (553-3001-379). The benefits derived from the NXFER feature include:

- minimal use of access TIE lines
- improved transmission performance, since TIE lines are not used for the completed connection
- operation identical to that of Call Transfer (XFER)

NXFER operates within the following parameters:

- Meridian 1 proprietary telephones must be equipped with a Call Transfer key.
- Network Signaling (NSIG) must be provided on both switches.

Network Signaling

Network Signaling (NSIG) provides a proprietary signaling protocol for transmission of network call information between switches that operate in a private network environment with Basic/Network Alternate Route Selection (BARS/NARS) or CDP. Network Signaling (NSIG) can be equipped at the Node and Main switches. For a detailed description of Network Signaling, refer to *Basic Network Features* (553-3001-379).

NSIG supports transmission or reception of information between the following switch types:

- System Node to System Node
- System Node to System Main
- System Node to an Electronic TIE Network (ETN) switch
- System Main to System Node
- ETN switch to System Node

Information transmitted and received from one switch to another can include the following:

- call type
- called number
- Network Class of Service (NCOS)
- Traveling Class of Service (TCOS)
- Traveling Class Mark (TCM)
- queue identification number (for CCBQ)

NSIG operates within the following parameters:

- A Main can connect to only one Node, and both switches must be equipped with the NSIG feature.
- TIE trunks between Nodes and Mains must be arranged for Dual-tone Multifrequency (DTMF) sending/receiving and wink-start operation.
- System Node compatibility with Electronic TIE Network (ETN) switches is limited to seven-digit on-network and ten-digit off-network calls.

Network Traffic

The Network Traffic (NTRF) feature enables traffic data related to BARS, NARS, and CDP to be retrieved and output at a traffic TTY. The network traffic measurements (in addition to the switch traffic measurements) are described in detail in *Traffic Measurement: Formats and Output* (553-3001-450).

Network Speed Call

Network Speed Call (NSC) enables a user who is normally restricted from making network calls to make such a call through BARS/NARS, provided that the destination is a number defined in a System Speed Call (SSC) list. When such a call is placed, the CLS and TGAR restrictions are lifted and a Network Class of Service (NCOS), associated with the SSC list, is assigned for the duration of the call. NSC is described in detail in *Basic Network Features* (553-3001-379).

Off Hook Queuing

Off Hook Queuing (OHQ) is an optional feature available at any switch equipped with BARS, NARS, or CDP. If all facilities are busy when an individual places a BARS, NARS, or CDP call, the OHQ feature enables the individual to wait off hook for a programmed length of time until a facility becomes available. OHQ is described in *Basic Network Features* (553-3001-379).

Operating parameters

Refer to the appropriate Nortel Networks technical publication for each ESN feature.

Feature interactions

Refer to *Electronic Switched Network: Signaling and Transmission Guidelines* (553-3001-180) for ESN feature interactions.

Feature packaging

Basic Authorization Code (BAUT) package 25 requires:

- Charge Account/Authorization Code (CAB) package 24.

Basic Alternate Route Selection (BARS) package 57 requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

Coordinated Dialing Plan (CDP) package 59 requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Flexible Call Back Queuing (FCBQ) package 61

Network Alternate Route Selection (NARS) package 58 requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

Network Authorization Code (NAUT) package 63 requires:

- Charge Account/Authorization Code (CAB) package 24
- Basic Authorization Code (BAUT) package 25

and at least one of the following:

- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58 or
- Coordinated Dialing Plan (CDP) package 59

Network Call Transfer (NXFR) package 67 requires:

- Network Class of Service (NCOS) package 32
- Network Signaling (NSIG) package 37

Network Signaling (NSIG) package 37 requires:

- Network Class of Service (NCOS) package 32

Network Traffic (NTRF) package 29 requires at least one of the following:

- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Priority Queuing (PQUE) package 60
- Flexible Call Back Queuing (FCBQ) package 61, or
- Off Hook Queuing (OHQ) package 62

Network Speed Call (NSC) package 39 requires:

- System Speed Call package (SSC) package 34

and at least one of the following:

- Basic Alternate Route Selection (BARS) package 57, or
- Network Alternate Route Selection (NARS) package 58

Off Hook Queuing (OHQ) package 62 requires

- Basic Queuing (BQUE) package 28

and at least one of the following:

- Basic Alternate Route Selection (BARS) package 57, or
- Network Alternate Route Selection (NARS) package 58

Feature implementation

Refer to *Electronic Switched Network: Signaling and Transmission Guidelines* (553-3001-180) for ESN implementation.

Feature operation

Refer to *Electronic Switched Network: Signaling and Transmission Guidelines* (553-3001-180) for ESN operation.

Emergency Services Access

Emergency Services Access (ESA) is a feature that places a customer in compliance with new federal legislation that requires the Private 911 type of functionality provided by ESA. Please note, however, that the ESA feature is also generally useful for users who are not subject to legislation, and is broad enough to be used in different countries. For example, it will be appreciated by any customer who wants to route emergency calls in a special manner, or who wants to be notified when a telephone user makes an emergency call. It would also appeal to a customer who wishes to have ESA calls answered on-site, on the business premises, rather than being forwarded to the Public Services Answering Point (PSAP).

Please refer to *Emergency Services Access: Description and Administration* (553-3001-313) for complete information.

End of Selection

Contents

This section contains information on the following topics:

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Feature operation 254

Feature description

This feature allows an End of Selection (EOS) signal to be sent back on a Direct Inward Dialing (DID) trunk to inform the Public Exchange/Central Office that the dialing phase of the call has been completed. The signal will be sent back when one of the following occurs:

- the DID call terminates on an idle station or attendant, an Automatic Call Distribution (ACD) queue, or a busy station
- the call has been intercepted (the DN is busy, not in service, or prohibited), and
- the interdigit timer has expired or an incomplete DN has been dialed.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the End of Selection feature described above.

The End of Selection feature is available with either the QPC357 or NTD9447 pack for analog trunks, or the QPC536 pack for 2 Mbit digital trunks. It is not available on 1.5 Mbit digital trunks or Japanese DMI trunks.

If the DN size is specified, the End of Selection feature allows a trunk to be locked out if the correct number of digits are not received, or if termination has not been completed when the correct number of digits have been received.

The End of Selection signal is not supported by R2 Multifrequency.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes:

Prompt	Response	Description
...		
EOS	(NO) YES	End of Selection (EOS) signal is enabled; no EOS signal. EOS and BSY signals are enabled.

Feature operation

No specific operating procedures are required to use this feature.

End of Selection Busy

Contents

This section contains information on the following topics:

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Feature operation 256

Feature description

This feature can be used where there is a requirement for the system to send a busy signal to the Public Exchange/Central Office when the call terminates in a busy connection. The signal will be sent 500 to 900 milliseconds after the end of selection signal is sent and informs the Central Office to release the connection and return busy tone to the originating source.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the End of Selection Busy feature described above.

The End of Selection Busy feature is only available on the NTD9447 or the QPC536 pack, and is not supported by R2 Multifrequency Compelled Signaling.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for trunk routes.

Prompt	Response	Description
...		
EOS	BSY	End of Selection (EOS) and BSY signals are enabled.

Feature operation

No specific operating procedures are required to use this feature.

End-of-Dialing on Direct Inward/Outward Dialing

Contents

This section contains information on the following topics:

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Feature operation 258

Feature description

This feature monitors an outgoing Direct Inward Dialing (DID) or Direct Outward Dialing (DOD) call to determine whether additional digits are dialed after the route access code seizes the trunk. If no digits are dialed in 15 seconds, the trunk is disconnected.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the End-of-dialing on DID/DOD feature described above.

The End-of-Dialing on DID/DOD feature is not available on 1.5 Mbit digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

No change to existing configuration is required for the End-of-dialing on Direct Inward/Outward Dialing feature.

Feature operation

No specific operating procedures are required to use this feature.

End-to-End Signaling

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This section contains information on the following topics:

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Feature description

The End-to-End Signaling (EES) feature enables a station to send Digitone end-to-end signaling through an established outgoing connection. EES provides fast reliable service and an optional feedback tone to the originator, as specified in LD 56. In addition, EES eliminates the use of a conference loop for sending EES tones to the connected parties.

To use EES, the following prompt or prompts need to be set in LD 15: EEST = NO (no feedback tone, default value) or EEST = YES, DTMF = NO (single optional feedback tone, as specified in LD 56).

An outgoing connection from a digital telephone is considered established after the end of dialing time is elapsed. Alternatively, an outgoing call can be established after the end of dialing time is elapsed, or can be established immediately by pressing an octothorpe (#) after the last digit is dialed.

Attendant End-to-End Signaling

The attendant can send DTMF tones to either the source or destination party using the AEES key on the Attendant Console. If there are two receiving parties on the current active loop key, the attendant can press the EXCL SRC or EXCL DEST key to exclude one of the connected parties before pressing the AEES key (defined in LD 12). Only one party on the active loop key (source or destination) can receive the DTMF signal. After pressing the AEES key, the attendant can press digits to send DTMF tones out to the source or destination party. To terminate the EES operation, the attendant should press the AEES key again. The states for the EXCL SRC, EXCL DEST, SRC loop, and DEST loop keys remain the same as before the EES key is pressed.

Operating parameters

The EES capability extends to internal analog (500/2500 type) telephone calls and incoming trunk calls.

A call must be established before using the EES feature. An outgoing call is considered established 14 seconds (DP trunk) or four seconds (2500-type telephone or Digitone trunk) after the last digit has been outpulsed. The length of this delay can be changed through service change. EES can be performed after end of dial time out, or when an answer supervision has been received from the far end, by pressing the octothorpe key (#) after the last digit.

EES is allowed only on CO, FEX, WATS, TIE, CCSA, DID, and CAMA trunk types.

EES is not available on analog (500/2500 type) telephones.

EES eliminates the use of the conference loop.

The AEES key, like other flexible programmable keys, cannot be configured on key 0 or key 1 of the Attendant Console.

There is a 5.4 dB difference between when EEST is set to YES (provide end-to-end signaling feedback tone) and when it is set to NO (provide no tone). An attenuation of 5.4 dB using the conference pads is applied to the EES tone if user feedback is to be given.

Feature interactions

End-to-End Signaling feature interactions

Agent/All Observe

In the Agent/All Observe mode, a supervisor, agent, and customer are all in a conference call. This feature uses Conference EES.

Attendant End-to-End Signaling

An Attendant Console in Attendant End-to-End Signaling mode can communicate with the source or destination party through in-band DTMF tones on an established speech path. The Attendant Console is treated like any other telephone.

Autodial Tandem Transfer

EES is used to send the Automatic Dialing (ADL) digits to the Public Exchange/Central Office (CO). With Autodial Tandem Transfer (ATX), the 911 agent can use the ADL key or manually dial the digits, or use a combination of both methods, to dial the third party's number. The ADL key can be pre-programmed with a prefix and the remaining digits can be dialed manually to distinguish between different numbers.

To get uniform feedback tone when using the ADL key along with manual dialing, set the DTMF prompt to NO in LD 15.

Call Modification

If EES is in progress, Call Modification is blocked. If Call Modification were not blocked, it might not be performed correctly during EES.

Call Detail Recording Record

An option in the Customer Data Block (LD 15) defines whether EES digits should be captured in the Call Detail Recording (CDR) record or not. This can prevent EES digits that contain sensitive information, like account numbers and passwords, from appearing in the CDR record.

Call Party Name Display

When entered after a call is answered, EES digits are displayed immediately following the CPND name of the connected party. Leading DN digits and name characters may be shifted out of the display window.

Conference End-to-End Signaling

Improved EES does not apply when the parties are in a conference call. In conference EES, a Tone and Digit Switch (TDS) loop is attached to the conference loop when a digit is pressed by one of the conferenced parties, and TDS is released when the digit is released. The setting of the EEST prompt determines whether the DTMF feedback tone is provided or not. The DTMF prompt is ignored for Conference EES.

EuroISDN Continuation

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

EuroISDN Trunk - Network Side

EuroISDN Master Mode

End-to-End Signaling, which allows in-band dialing to be performed on ISDN trunks before and after the call has been answered, is supported on the EuroISDN Trunk - Network Side connectivity.

In the case of tandem with ISDN trunks, the necessary information to allow the End-to-End Signaling feature is tandemed to the ISDN trunk. At this point, it becomes the responsibility of the end user switch to provide the End-to-End Signaling service.

Multi-Party Operations – Three-Party Service

The party receiving the patience tone or the Misoperation ringback is not able to use EES.

Silent Observe

EES supports the Silent Observe feature of Automatic Call Distribution (ACD), like any other feature that involves EES between two telephones. A supervisor can use this ACD feature to silently observe an agent.

Stored Number Redial

End-to-End Signaling (EES) activates after a call to a trunk is established by expiration of the end-of-dial timer. Further digits dialed are not stored by the SNR feature once it is in EES mode.

Attendant End-to-End Signaling feature interactions

Attendant Administration

While in the Attendant Administration mode, pressing the AEES key is ignored.

Attendant Barge-In
Attendant Busy Verify

While in the Barge-In/Busy Verify mode, the console cannot enter AEES mode.

Attendant Features

Activating Automatic Wake Up, Call Park, Charge Account, Calling Party Number, Hold, Release, or another loop key will terminate AEES operation.

Attendant Position Busy
Centralized Attendant Service
Night Service

These features work together with Attendant End-to-End Signaling (AEES). However, do not press one of these feature keys while using AEES, or the Dual-tone Multifrequency (DTMF) code signals may be blocked.

Attendant Supervisory Console

The supervisor can operate AEES if there is a call on the active loop key. An attendant in AEES mode can be monitored by the supervisor.

Conference

While in AEES mode, the receiving party cannot initiate a conference call.

End-to-End Signaling (station level)

The Attendant Console and the telephone receiving AEES cannot both activate EES simultaneously.

Interposition call

When an attendant is actively connected to another console using Interposition Attendant Call, AEES is blocked. During an Interposition Call Transfer, however, the console that is actively connected to a telephone can perform AEES, provided the party connected to the other Attendant Console is excluded.

Meridian Hospitality Voice Services – Digit Key

Attendant End-to-End Signaling and Digit Key are mutually exclusive. Being in AEES mode overrides the use of the Digit Key.

Trunk connection

On incoming ground start CO or Direct Inward Dialing (DID) trunks without Answer Supervision, you must press the Release (RLS) key on the console to exit AEES mode and drop the connection.

Feature packaging

End-to-End Signaling and Attendant End-to-End Signaling are both part of package 10 and have no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable End-to-End Signaling tone feedback.

2 LD 12 – Add End-to-End Signaling key to Attendant Console.

3 LD 56 – Specify the cadence for the EES feedback tone.

LD 15 – Enable End-to-End Signaling tone feedback.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- EEST	(NO) YES	NO = No EES feedback tone is given to the telephone. YES = EES feedback tone is given; the type is defined by the DTMF prompt. For Small Systems, DTMF should be set to NO.
- DTMF	(NO) YES	NO = Use EES for single feedback tone. YES = Use EES for DTMF feedback tone. For Small Systems, DTMF should be set to NO.
...		
TYPE	CDR	CDR and charge account options.
- ECDR	(NO) YES	NO = Do not capture EES digits in the CDR record. YES = Capture EES digits in the CDR record.

LD 12 – Add End-to-End Signaling key to Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx EES	Add EES key (xx = key number) (cannot be key 0 or 1).

LD 56 – Specify the cadence for the EES feedback tone.

Prompt	Response	Description
REQ	CHG NEW	Change, or add.
TYPE	FTC	Flexible Tones and Cadences.
TABL	x	FTC table number.
HCCT	YES	Hardware Controlled Cadence.
EEST		No response expected; this is an informational prompt.
- TDSH	i bb cc tt	TDS external, burst, cadence, and tone.
- XTON	0-255	NT8D17 TDS tone code.
- XCAD	0-255	NT8D17 cadence code for FCAD.

Feature operation

No specific operating procedures are required to use this feature.

End-to-End Signaling Display Enhancement

Contents

This section contains information on the following topics:

Feature description 267
Operating parameters 268
Feature interactions 268
Feature packaging 269
Feature implementation 270
Feature operation 270

Feature description

The End-to-End Signaling Display Enhancement (EESDSP) feature enhances the existing End-to-End Signaling (EES) feature. EES digits can communicate private information such as account numbers, authorization codes, and passwords. In some environments, showing this information can be a security issue. EESDSP feature provides the option to show or block the EES digits from appearing on a set's display screen. The customer can enable or disable this option at the EES Digit Display (EESD) prompt in the Customer Data Block.

With the EESDSP feature enabled, the user's display shows all the EES digits as dialed. EES digits display when you enter them after a call is answered. The digits appear following the Call Party Name Display (CPND) name of the connected party. Initial digits and name characters may move out of the display window if necessary. With the EESDSP feature disabled, the user's display does not change, keeping the established call information.

Operating parameters

The EES feature must be enabled for the EESDSP feature to function.

The EESDSP feature applies only to the EES digit display functionality of the existing EES features. The EES digits are not displayed on the sets of the other parties in an established call.

The EESDSP feature does not apply to a networking environment.

The EESDSP feature applies to proprietary sets, Basic Rate Interface (BRI) sets, and Attendant Consoles with a display screen enabled to show entered EES digits and EES capabilities.

Attendant Consoles require Attendant EES (AEES), which is enabled by configuring and using the programmable AEES key.

Feature interactions

The EESDSP feature does not change the production of tones for EES digits, or the processing or sending of EES digits. This feature only gives the customer the option to show or block all EES digits on the display.

Attendant End-to-End Signaling

For Attendant End-to-End Signaling (AEES), place the Attendant Console in EES mode by pressing the AEES key. When in EES mode, you can dial EES digits. The Attendant Console can send the EES Dual-tone Multifrequency (DTMF) tones to either the source or destination party.

When the End-to-End Signaling Display Enhancement option is enabled, the Attendant Console display shows the EES digits entered while in the EES mode. For QCW4 type Attendant Consoles, the digits appear on the one line display. For M1250 and M2250 type Attendant Consoles, the digits appear on the second line of the display. If disabled, the Attendant Console display does not change.

Call Party Name Display (CPND)

With the EESDSP option enabled, EES digits appear after the Call Party Name Display (CPND) name of the connected party. Initial digits and name characters may move out of the display window if necessary.

With the EESDSP option disabled, the set display does not change from the established CPND display.

Conference End-to-End Signaling

The EESDSP option changes the display of the EES digits as dialed for all the EES features, including Conference EES.

End-to-End Signaling

The EESDSP option has no effect on the digits dialed before the system is in EES mode. In EES mode, digits dialed from a set with a digital display appear on the display when the EESDSP option is enabled. When you disable the EESDSP option, the display does not show the dialed EES digits.

Improved End-to-End Signaling

The EESDSP feature changes the display of EES digits the same for both Improved End-to-End Signaling (IEES) and EES.

Feature packaging

The End-to-End Signaling Display Enhancement (EESDSP) feature requires End-to-End Signaling (EES) package 10.

Feature implementation

LD 15 – Enable the End-to-End Signaling Display Enhancement feature.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Customer Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
....	....	
EEST	 (NO) YES	EES Tone to originating party. Do not send feedback to the originator. Send feedback tone to the originator. Enhanced EES signaling is provided when EEST=YES and DTMF=NO.
- DTMF	 (YES) NO	EES feedback tone. EES for DTMF feedback tone. EES for single tone feedback (only prompted if EEST=YES).
EESD	 (NO) YES	EES digit display. Do not display the EES digits. Display all EES digits.
TTBL	(0)–31	Tone Table number.
....	....	

Feature operation

No specific operating procedures are required to use this feature.

Equal Access Compliance

Contents

This section contains information on the following topics:

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Feature description

A telephone user can select any interexchange carrier for any given call by using a Carrier Access Code (CAC). A CAC comprises an Equal Access identifier and a Carrier Identification Code (CIC). Nortel Networks refers to a call preceded by a CAC as an Equal Access call.

Federal Communications Commission (FCC) requirements

FCC Part 68 regulations require that any equipment or software manufactured or imported on or after April 17, 1992, and installed by any aggregator, must allow all users to use Equal Access codes to selectively access the long distance carrier of their choice. As defined in FCC docket 90-313, an *aggregator* is any business that, in the ordinary course of operations, makes telephones available to the public, or to transient users of the premises, for interstate telephone calls using a provider of operator services. Aggregators include hotels or motels, hospitals, universities, airports, gas stations, or pay telephone owners.

Aggregators, although they must allow callers access to any long distance caller, are permitted to block calls selectively. Selective equal access lets aggregators choose to block direct-dialed calls that result in charges to the originating telephone. Aggregators cannot block operator-assisted calls.

Nortel Networks complies with the FCC Equal Access rules in dockets 90-313, 91-35, and their appendixes.

Equal Access dialing plans

The system software supports Equal Access dialing plans as follows:

- It allows operator-assisted North American and international dialing.
  - CAC + 0
  - CAC + 0 + (NPA) + NXX + XXXX, and
  - CAC + 01 + CC + NN.
- It allows or denies direct North American and international dialing.
  - CAC + 1 + (NPA) + NXX + XXXX, and
  - CAC + 011 + CC + NN.

Legend:

CAC = Carrier Access Code (101XXXX)

NPA = Numbering Plan Area (area code in the North American Numbering Plan)

NXX = end-office code
(N = any digit except 0 or 1; X = any digit (0–9))

XXXX = any four digits

CC = Country Code

NN = National Number

Route types

Equal Access Compliance supports COT, FEX, WAT, DID, and TIE routes.

A TIE route is supported only if standard signaling is specified in LD 16 (SIGO = STD). To enable Equal Access call restrictions to function properly, Digital Trunk Interface (DTI) TIE routes must be voice only. (DTI TIE routes configured as voice/data are not supported for connection to a Public Exchange/Central Office.) TIE routes must be either outgoing or incoming/outgoing (ICOG = IAO or OGT).

Call restriction

Call restriction relies on fixed pattern recognition to determine which calls can be denied. Switch administrators can restrict two kinds of direct-dialed Equal Access calls: North American calls with the 101XXXX+1+NPA+NXX+XXXX format and international calls with the 101XXXX+011+CC+NN format. If either restriction option is chosen, the administration must verify that the Original Carrier Access Code (OCAC) flag is correctly set.

Call restrictions do not affect attendant calls.

Calls blocked by Equal Access are not directed to alternate routes.

BARS/NARS routing

Equal Access determines restrictions without looking at a call's originating type (ESN or Direct Access). Routing has no effect on Equal Access call restriction: calls receive the same restriction treatment whether they originate from a trunk access code or from BARS/NARS. Equal Access is not a BARS/NARS feature and does not require BARS/NARS dialing.

To configure BARS/NARS to route Equal Access calls, simply use a special number (SPN) of 10 (the Equal Access code) to identify the calls as Equal Access calls and route them accordingly.

Example

Configure BARS/NARS for Equal Access call routing, assuming that calls originate from Customer 0 and go out over Route 10. To route Equal Access calls originating from Customer 0 over Route 10, using route list index 100 and access code 1 (AC1).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Equal Access compliance is included in base system software. Network Class of Service (NCOS) package 32 is required to configure Equal Access.

Feature implementation

The configuration in this example routes all Equal Access calls placed through BARS/NARS with access code 1 (AC1) over route 10. Set the SPN to "101".

In this example set Equal Access toll calls for NCOS = 4. Note that Equal Access toll calls placed through direct trunk access to route 10 also will be blocked.

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Set OCAC as appropriate.

2 LD 86 – Set the route list index to Route 10.

3 LD 90 – Establish an SPN for the Equal Access code.

4 LD 87 – Configure a NCOS for Equal Access.

5 LD 10 – Assign a NCOS to an Analog Telephone.

6 LD 11 – Assign a NCOS to a Digital Telephone.

7 LD 16 – Enable Equal Access for this route.

Carrier Identification Code Expansion supports and extends the General Carrier Restriction method of blocking calls. Given the expansion in the number of Carrier Identification Codes (CIC), it is no longer practical to support Selective Carrier Restriction functionality. Carrier Identification Code Expansion continues to provide the selective blocking function required by the FCC; Nortel Networks and the FCC interpret the term “selective” differently. For these reasons, prompts pertaining to General Carrier Restriction and Selective Carrier Restriction in LD 16 no longer appear.

Customers who chose the ITOL prompt in LD 16 to block international calls should also have international calls blocked at the Public Exchange/Central Office to reduce the likelihood of unauthorized international calls. The carrier or Central Office operator intercept interdigit timer typically expires in four to six seconds. The system end of dial timers, End-of Dial Timer for non-Digitone Trunks (EOD) and End-of Dial-Timer for Digitone Trunks (ODT), are defaulted to 14 and four seconds respectively. ODT can be raised to seven seconds to prevent Digitone stations from bypassing Equal Access restrictions of Digital Distance Dialing international calls.

The interdigit timeout for non-leftwise-unique prefixes 0 and 01 is fixed for a given carrier network. Therefore, Equal Access connects the call to the Central Office trunk if the user dials Carrier Access Code + 0 and allows the end-of-dialing timer to expire. Equal Access blocks the same call if the caller presses the octothorpe (#) key and cancels the EOD or ODT. The caller cannot bypass the EQAR prompt in LD 16 provided that the EOD and ODT are set long enough to exceed the inter-digit timeout on the carrier networks.

Before and during the permissive period, when both the three-character and the four-character CIC are allowed, current Equal Access users must set the Original Carrier Access Code (OCAC) flag to YES in LD 17. OCAC should be set to NO (default).

New Equal Access customers do not need to change the OCAC flag until the feature is configured.

LD 17 – Set OCAC as appropriate.

Prompt	Response	Description
REQ	CHG	Change existing route data.
TYPE	CFN PARM	Configuration Record. System parameters.
PARM	YES	Change system parameters.
- NDRG	(NO) YES	(Disable) enable new distinctive ringing.
- OCAC	(NO) YES	Support original CAC format (must be set to YES during interim period, NO following interim period).

LD 86 – Set the route list index to Route 10.

Prompt	Response	Description
REQ	NEW CHG	Create, or change database.
CUST	xx	Customer number, as defined in LD 15
FEAT	RLB	Route List Block.

RLI	100	Use route list index 100 to route Equal Access calls.
ENTR	0	Route entry number for this route list index (0 if this is the first entry).
ROUT	10	Send Equal Access calls over Route 10.

LD 90 – Establish an SPN for the Equal Access code.

Prompt	Response	Description
REQ	NEW	New ESN translation table entry.
CUST	xx	Customer number, as defined in LD 15
FEAT	NET	Network translation table entry.
TRAN	AC1	Access code 1 is used to originate the Equal Access calls.
TYPE	SPN	SPN translation entry.
SPN	101	SPN (Equal Access code).
RLI	100	Use route list index 100 to route Equal Access calls.

LD 87 – Configure a NCOS for Equal Access.

Prompt	Response	Description
REQ	CHG	Change NCTL data.
CUST	xx	Customer number, as defined in LD 15
FEAT	NCTL	Change NCTL block.
NCOS	4	Network Class of Service group number.
EQA	YES	This NCOS permits Equal Access call restriction capabilities.

LD 10 – Assign a NCOS to an Analog Telephone.

Prompt	Response	Description
REQ:	CHG	Change existing set data.
TYPE:	aaa	Specify set type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
NCOS	4	Network Class of Service group number.

LD 11 – Assign a NCOS to a Digital Telephone.

Prompt	Response	Description
REQ:	CHG	Change existing set data.
TYPE:	aaa	Specify set type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
NCOS	4	Network Class of Service group number.

LD 16 – Enable Equal Access for this route.

Prompt	Response	Description
REQ	CHG	Change existing route data.
TYPE	RDB	Change Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	10	
EQAR	(NO) YES	Enter YES to enable Equal Access and selective blocking for this route. A YES response triggers the next two prompts.

- NTOL	(DENY) ALOW	Specify that Equal Access North American calls billed to originating telephone are to be denied.
- ITOL	(DENY) ALOW	Specify that Equal Access international calls billed to originating telephone are to be denied.

Feature operation

No specific operating procedures are required to use this feature.

Extended DID/DOD Software Support – Europe

Contents

This section contains information on the following topics:

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Feature interactions 286
Feature packaging 288
Feature implementation 288
Feature operation 291

Feature description

This feature provides software support for the European Extended Direct Inward Dialing (XDID)/Direct Outward Dialing (DOD) cards. These cards are the NT5K36AA (German XDID pack), NT5K84AA (Swiss XDID pack) and NTAG04AA (Dutch XCOT/DID). The new packs enable the system to have the following Intelligent Peripheral Equipment (IPE) DID/DOD functionalities.

Seizure acknowledgment on outgoing traffic

In order to provide this functionality, LD 14 has to be modified in order to allow Ear and Mouth (E&M) signaling to be configured for DID trunk on an XDID card. A new "Trunk Type and Signaling" in the type 2 Channel Download message defined for DID-E&M has to be downloaded onto the firmware. LD 14 must have a configuration of ACWK = YES, and LD 16 must have a configuration of "trunk type" (TYPE) = DID, "signaling" (SIGL) = EAM, and "start arrangement on outgoing" (STRO) = IMM.

End of dialing on DOD

No software changes are required to provide this functionality.

Interdigit timer on DID

To provide this functionality, the Partial Dial (PRDL) prompt has to be configured as BSY or YES.

End-of-selection signal on DID

To provide this functionality, a new outgoing SSD message, "End of Signaling", has been defined. The End-of-selection (EOS) prompt in LD 16 has to be configured to BSY or YES.

End-of-selection busy signal on DID

To provide this functionality, a new Outgoing SSD message "End of Signaling Busy", has been defined. The End of Selection (EOS) prompt in LD 16 has to be configured to BSY.

Provision of busy tone, ringback tone, and overflow tone for DID callers

No software changes are required to provide this functionality.

Restricted/unrestricted DID Class of Service for DID calls

No software changes are required to provide this functionality.

DID to TIE connection, subject to configured trunk barring and Class of Service restrictions

To provide this functionality, the DITI prompt in LD 15 has to be configured to YES.

Line Break Alarm

To provide this functionality, an incoming SSD message, BAR, has been defined to trigger the Trunk Failure Monitor feature whenever a problem situation arises on the line. A new SSD message, UNBAR, has been defined to clear the problem indications provided by the Trunk Monitor feature. LD 14 has to be configured with a Class of Service of trunk barring allowed (BARA) or denied (BARD). This Class of Service is downloaded onto the XDID/DOD cards.

Static loss pad

One of two loss pads (either long or short) can be selected on a per trunk basis. To provide this functionality, LD 14 has to be configured with a Class of Service of either SHL (short line) or LOL (long line). The configured pad type is downloaded onto the XDID/DOD cards.

Disconnect supervision

To support this functionality, the software has been changed so that an XDID card can provide disconnect supervision for a DID trunk with Ear and Mouth (E&M) signaling. The software has also been changed to refrain from sending an End of Selection (EOS) signal when an incoming trunk call is being disconnected.

DID digit collection type

To support this functionality, the type of incoming DID digit collection is configured against a Class of Service and downloaded to the XDID card.

Unsupported Class of Service

If an attempt is made to download an unsupported configuration during regular enabling of the pack or during audit, the pack responds with a problem report type 3 message. The error message ERR5327 is printed out on the TTY and the trunk is disabled.

Incoming Digit collection

This functionality only applies to Dual-tone Multifrequency (DTMF) DID trunks. The software must be ready to accept incoming digits regardless of whether or not an “Enable Digit Collection” message is sent. To support this functionality, the trunk must be configured with an incoming start arrangement (prompt STRI = IMM in LD 14). Message H0019 is sent when a Digitone Receiver (DTR) signal is found.

Proceed to Send message to the firmware

A “Proceed to Send” message must be sent to the firmware in cases of non dial pulse trunks, as soon as the software is able to receive digits. To support this functionality, LD 14 must be configured with DTCR = YES. A new H0019 message is sent when a Digitone Receiver (DTR) signal is found for Dual-tone Multifrequency (DTMF) signaling, or when a Multifrequency Compelled (MFC) sender/receiver is found for Multifrequency Compelled (MFC) signaling. If a DTR signal is not found, the call is released.

PPM and Buffered PPM downloadable on a per country basis

To support this functionality, Periodic Pulse Metering and Buffered Periodic Pulse Metering (PPM) are enabled on a per trunk basis, rather than on per card basis. Configuration of PPM and Buffered PPM is still done on a per route basis.

Audit conflict reporting and PPM event reporting

To support this functionality, a channel and card parameter download audit is performed during initialization and when LD 30 is run as a midnight routine. This is to ensure that the software configuration is the same as the configuration stored in the hardware. If a discrepancy is detected, the software information is stored in the hardware and an error message is printed on the TTY. Also, for PPM recording, two new type 5 messages have been defined to report hardware problems. These are the TRK Event: Partial Metering Detection Failure message and the TRK Event: Fatal Metering Detection Failure message.

On partial PPM failure, a TRK516 error message is printed on the TTY. If PPM is configured, CDR records for any calls in progress may be incorrect. If Busy Tone Supervision is configured, busy tone may not have been detected for calls in progress. On fatal PPM failure, a TRK517 error message is printed on the TTY. If PPM is configured, further PPM reporting is disabled until the pack is either disabled and then reenabled, or removed and then reinserted. The CDR record for any call in progress is incorrect. If Busy Tone Supervision is configured, tone supervision can no longer be performed until the pack is either disabled and then reenabled, or replaced.

Network DID and Enhanced Night Service groups on DID

No software changes are required to support these functionalities on the XDID/DOD cards.

Held call clearing

No software changes are required to support this functionality on the XDID/DOD cards.

Unequipped channel notification

To support this functionality, a channel download message is sent to the XDID pack whenever a trunk on the pack is removed.

Call blocking

Before disabling a trunk, the software requires confirmation that the trunk is in the idle state. To support this functionality, the software disable sequence has been modified. The software waits for an idle state message from the XDID pack before sending a disable message to the trunk. If the idle message is not received before the disconnect supervision (DSI) timer expires, the software prints the TRK136 (Release Failure on the Unit) error message. The trunk is placed in lockout state. If the disable sequence was started from an overlay, a TRK520 (No Far End Release) error message is printed. The trunk remains in lockout until a Far End Release message is received on the pack.

Number Reception message

This is a Dutch Central Office (CO) requirement. When sufficient digits are received at the Dutch CO, the battery is reversed. When the Dutch COT/DID pack (NTAG04AA) detects this reversal, it sends a Number Reception message. This functionality is a software enhancement.

When an ENLC command is performed on an XDID/E&M card, the card is first reset and then messages are downloaded to the firmware to reflect the software trunk state. This prevents the software database from being in conflict with the firmware database. If an XDID/E&M card unit is in busy state, the SSD message H.A004 is printed. If the unit is in barred state, the SSD message H.A003 is printed.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

DID/DOD

This feature provides the same feature interactions as the following DID/DOD features:

- End of Selection, End of Selection Busy
- Provision of Tones

- Selectable DN Size
- Partial Dial Timing
- Seizure Acknowledgment
- DID Restricted Class of Service
- DID to TIE Connection, and
- Enhanced Night Service.

Japanese DID trunk

For Japanese DID trunk support, DID to TIE (DTOT) package 176 must be removed due to tariff restrictions.

Federal Communications Commission (FCC) Compliance for DID Answer Supervision

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XDOD units, it may lead to incorrect call status. Therefore, equipping this package is not recommended.

Trunk Failure Monitor

When a BAR message indicating a problem situation is received, a TRK501 message is printed on the TTY, the uppermost key lamps light up on the Attendant Console, and the trunk is placed in the BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A TRK502 message is printed on the TTY, the lamps on the Attendant Console are darkened, and the trunk is idled.

> ***Note:*** BARA CLS must be configured on the XDID trunk for the described process to occur.

XDID/DOD and XFCOT

Software support for European XDID/DOD cards and software support for European XFCOT cards provide similar functionality in the following areas:

- Trunk Failure Monitor processing
- Downloading of PPM information

- Configuration and downloading of static pad setting for short line and long line, and
- Configuration download processing. Fields that are not filled due to configuration limitations are left blank and are not validated or interpreted by the firmware. The fields are treated as unused fields.

The DTCR (Digit Collection Ready) prompt has replaced the DTRA (Digitone Receiver Attached) prompt in LD 14.

Feature packaging

The Extended DID/DOD Software Support feature requires the following packages:

- Meridian 1 Superloop Administration (XCT1) package 205, which has the following requirements:
  - Intelligent Peripheral Equipment (XPE) package 203
  - International Supplementary Features (SUPP) package 131
  - ISDN Supplementary Features (ISDNS) package 161
  - PPM/Message Registration (MR) package 101
  - Trunk Failure Monitor (TFM) package 182.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – In the Customer Data Block, allow DID to TIE connections.

2 LD 16 – Define a DID/DOD trunk route for Germany and Switzerland.

3 LD 14 – Define an XDID card unit.

LD 15 – In the Customer Data Block, allow DID to TIE connections.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	NET	ISDN and ESN Networking options
...		
- DITI	YES	DID to TIE connections are allowed.

LD 16 – Define a DID/DOD trunk route for Germany and Switzerland.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
TYPE	RDB	Route Data Block
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	 DID	Trunk Type Direct Inward Dial
ICOG	 IAO	Incoming Outgoing trunk Incoming And Outgoing
ACOD	xxxxxxx	Access Code for the trunk route
CNTL	YES	Change Controls or timers
TIMR		Trunk Timers
	EOD 10112	End-of-Dial timer
	GTI 128	Incoming Guard timer
	GTO 128	Outgoing Guard timer

	ICF 0	Incoming Flash timer
	OGF 0	Outgoing Flash timer
	DSI 360000	Disconnect Supervision timer
NEDC		Near End Disconnect Control
	ETH	Either end control
FEDC		Far End Disconnect Control
	ETH	Either end control
...		
MR		Message Registration
	PPM	Buffered PPM signals to be counted on this route
	XLD	M&MM Lead non-buffered is used.
PRDL		Partial Dial timing
	BSY	Busy signal is sent on timeout
EOS		End Of Selection signal
	BSY	EOS and busy signals are enabled
ACKW	(NO) YES	Acknowledgement seizure signal
BTT	100	Busy Tone Time

LD 14 – Define an XDID card unit.

Prompt	**Response**	**Description**
REQ	NEW	Create a New Data Block.
	CHG	Change an existing Data Block.
TYPE	DID	Direct Inward Dial trunk data block.
...		
XTRK	XDID	Extended (Intelligent Peripheral Equipment [IPE]) Direct Inward Dialing trunk.
...		

SIGL	EAM	Ear And Mouth (E&M) signaling (note that this prompt uses the letter “A”, instead of the “&” which is more commonly used in the abbreviation of Ear and Mouth).
STRI	IMM	Immediate Start arrangement Incoming.
STRO	IMM	Immediate Start arrangement Outgoing.
...		
CLS		Class of Service. The Class of Service parameters to be downloaded onto the XDID card unit.
	(LOL) SHL	Enter (LOL for long line) or SHL for short line static loss pad selection.
	(BARD) BARA	Barring (Denied) Allowed.
...		
DTCR	(NO) YES	Digit Collection Ready. Incoming digit collection ready; (do not) send acknowledgment when digit collection resources (DTR, MFC sender/receiver) are ready and attached.

Feature operation

No specific operating procedures are required to use this feature.

Extended Flexible Central Office Trunk Software Support

Contents

This section contains information on the following topics:

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Feature operation 303

Feature description

This feature provides software support for the following Extended Flexible Central Office Trunk (XFCOT) cards to meet the requirements of the following countries:

- NT5K70AA (German 8D)
- NT5K71AA (German 4D)
- NT5K82AA (Swiss)
- NT5K90AA (Danish PPM)
- NT5K90BA (Danish non-PPM)
- NT5K93AA (Norwegian PPM)

- NT5K93BA (Norwegian non-PPM)
- NTAG03AA (Dutch COT)
- NTAG04AA (Dutch DID/COT)
- NT5K18BA (New Zealand)
- NT5K99AA (Spanish PPM), and
- NT5K99BA (Spanish non-PPM).

The NT5K18AA (UK XFCOT) is not affected by the software changes introduced to support the XFCOT packs.

The following supervision, based on loop start signaling, is supported:

Battery Supervision Central Office Trunk (COT)

A battery supervised COT uses polarity detectors to provide seize, answer and disconnect supervision on all outgoing calls, and disconnect supervision on incoming calls. The supervision is performed by reversing the polarity from the Public Switched Telephone Network (PSTN) line. The battery supervised COT is configured in LD 14 with BAT.

ARF Supervision Central Office Trunk

ARF is an Ericson type series Public Exchange which provides disconnect supervision on both incoming and outgoing loop start Central Office trunk calls; on outgoing calls, seize supervision is also provided. Supervision is based on battery reversal detection. The signaling used to provide this supervision is called ARF signaling. The ARF supervised COT is configured in LD 14 with ARF.

Tone Supervised Central Office Trunk with downloadable Busy Tone parameters

A tone supervised COT has a busy tone detector on each unit. Busy tone is provided by the PSTN when the far end releases from outgoing and incoming trunks. The tone supervised COT is configured in LD 14 with BTS. This tone supervision depends on the busy tone frequency and cadence characteristics, as configured on a card basis using the Busy Tone ID (BTID) prompt in LD 14.

Loop Break Supervised Central Office Trunk

This type of signaling provides disconnect supervision by detecting a calibrated battery removal from the PSTN. The loop break COT supervision is configured in LD 14 with LBS.

Unsupervised Central Office Trunk

An unsupervised COT has neither polarity, battery, nor busy tone detector. Thus, no answer or disconnect supervision is provided for incoming or outgoing calls. A trunk is configured as unsupervised in LD 14 using other than BAT, LBS, ARF, or BTS.

Autoguard

Autoguard provides seize supervision on outgoing trunk calls. Autoguard is configured in LD 14 with SEIZ = YES.

Extended Flexible Central Office Trunk Software Support also provides the following capabilities:

- Trunk Barring. The XFCOT card can detect signaling from the PSTN that a trunk is barred, and that any call on the trunk must be dropped. The trunk unit is then marked software busy (busy barred) so that no outgoing calls may be made. A TRK514 message is printed on the TTY. A STAT (status) command in LD 32 or 36 yields a “Busy Barred” status. When the PSTN signals that the trunk unit may be unbarred, the software idles the trunk unit and a TRK515 message is printed on the TTY. Barring is configured on a per unit basis in LD 14 against a CLS of BARA. The BARA CLS is downloaded onto the XFCOT card.

- Static Loss Pad Selection. Trunk pad selection controls transmission loss. A pad may be inserted within or outside an XFCOT trunk card to allow a call to terminate on a station or another trunk. Two pad types are available to support long line or short line. The pad types are configured in LD 14 on a per unit basis, against a Class of Service of SHL for short line or a Class of Service of LOL for long line. The SHL or LOL is downloaded onto the XFCOT card.
- Enabling and disabling of Periodic Pulse Metering (PPM). The user configures PPM on a per route basis; the software configures the trunk on a per unit basis.
- Enabling and disabling of Buffered Periodic Pulse Metering (PPM), on a per trunk basis.
- A PPM ID that designates PPM parameters. This is configured in LD 14 against the PPID prompt. This value is downloaded onto the XFCOT firmware so that the appropriate PPM parameter may be selected.
- A four-unit card. The NT5K71AA four-unit quad density card has been introduced to meet German requirements.
- Mixed Central Office Trunk and Direct Inward Dialing on the same XFCOT card. In LD 14, the XFCOT card may be configured as being either COT or DID.
- ALS signaling. ALS, available only on the NTAG04AA (Dutch DID/COT) unit, is combined COT/DID signaling. Additions to the ground start signaling have been added for XFCOT support. On the near end, a partial release message is sent instead of a full release message. On an outgoing call, the number reception is accepted and interpreted by the software. Number reception is a battery reversal signaling from the CO, indicating that it has received sufficient dialing information. The ALS signaling type is configured in LD 14 against the SIGL prompt.

- Balance impedance adjustment. It is possible to download the balance and termination impedance configured by a craftsperson for a NT5K90AA (Danish PPM) or NT5K90BA (Danish non-PPM) unit. The termination impedance is defaulted to value of 600 ohms. The balance impedance may be configured in LD 14 using the BIMP prompt, as 600 ohms or 3COM (three-component).
- Flash hook signaling, for features requiring a flash hook operation. The flash hook signal instructs a pack to send a flash hook signal to the PSTN. The features that require a flash hook are Malicious Call Trace and Centrex Switchhook Flash.

Error reporting and auditing is also provided. New problem reports are defined so that the XFCOT card can notify the software when the dialing speeds or companding laws are not supported by the hardware. If these new error reports are received from the XFCOT card, an error message is printed on the TTY. A channel and card parameter download audit is performed during initialization and when LD 30 is run as a midnight routine to ensure that the software configuration is the same as the configuration stored in the hardware. If a discrepancy is detected, the software information is stored in the hardware, and an error message is printed on the TTY.

For PPM recording, two type 5 messages have been defined to report hardware problems. These are the TRK Event: Partial Metering Detection Failure message, and the TRK Event: Fatal Metering Detection Failure message. Also, a type 12 channel configuration message and a type 13 channel audit configuration message have been introduced. The type 12 message provides the hardware with certain card configuration information, so that the card may be able to inform the software when certain configurations are not supported on the pack, and perform message filtering based on the software configuration. The type 13 message provides configuration download messages during the midnight routine.

The following table summarizes the downloaded software configurations that each XFCOT card supports.

Table 5
Downloaded configurations for XFCOT cards

XFCOT card	Hardware I.D. supported	Signaling supported	Downloaded SUPN supported	Periodic Pulse Metering (PPM)
NT5K18AA	01, 13, 14	COT (GRD, LGR, LDC)	SUPN	per pack
NT5K16BA	00, 01	COT (LOP, GRD)	BTS	per pack & unit
NT5K70AA	00	COT (LOP)	BTS	per unit
NT5K71AA	00	COT (LOP)	BTS	per unit
NT5K82AA	00	COT (LOP)	BTS, LBS, BAR	per pack & unit
NT5K90AA	00	COT (LOP)	BTS, ARF	per unit
NT5K90BA	00	COT (LOP)		none
NT5K93AA	00	COT (LOP)	BTS	per pack & unit
NT5K93BA	00	COT (LOP)	BTS	none
NT5K99AA	00	COT (LOP)	BTS	per unit
NTAG03AA	00	COT (LOP)	BTS	per pack & unit
NTAG04AA	26, 27	COT (ALS) DID (EAM)		per pack & unit

Operating parameters

The flash hook implementation for the Centrex Switchhook Flash feature does not provide flexible timing, as is provided by non-XFCOT packs. The timing is hard-coded onto the pack at 90 milliseconds.

The new XFCOT trunks cannot support the PPM frequency characteristics, configured as the PPM ID, for each trunk. The PPM ID is configured for the first trunk configured for the pack, and cannot be changed unless all trunks are removed from the pack and then reconfigured. The same restrictions apply to the busy tone indication ID.

Only static pad selection is supported on the new XFCOT cards. Pad selection on a per call or per event basis is not supported.

Loop Start Supervisory Trunks and Japanese Central Office Trunks are not supported on the new XFCOT cards.

The B34 Codec support is not provided by this feature. The B34 Codec configured on a card allows the software to download an actual loop value for pads, rather than long line or shot line notations.

Periodic Clearing is not supported on the new XFCOT cards.

Feature interactions

Dial Tone Detector

A Dial Tone Detector notifies the software that a dial tone has been received for an outgoing call. With the XFCOT cards, dial tone detection is not attempted until a SEIZE ACKNOWLEDGE signal is received for those supervisions that require such a signal.

European XDID/DOD

Software support for European XDID/DOD cards and software support for European XFCOT cards provide similar functionality in the following areas:

- Trunk Failure Monitor or barring
- Downloading of PPM information
- Configuration and downloading of static pad setting for short line and long line, and
- Configuration download processing. Fields that are not filled due to configuration limitations are left blank and are not validated or interpreted by the firmware. The fields are treated as unused fields.

Federal Communications Commission (FCC) Compliance for DID Answer Supervision

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XFCOT units, it may lead to incorrect call status. Therefore, equipping the FCC package is not recommended.

Trunk Failure Monitor

When a BAR message indicating a problem situation is received, a trunk message is printed on the TTY, the uppermost key lamps light up on the Attendant Console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A message is printed on the TTY, the lamps on the Attendant Console are darkened, and the seized trunk is idled. Note that BARA Class of Service must be configured on the trunk for the described processing to occur.

UK XFCOT (NT5K18AA)

For the UK XFCOT card, the NT5K18AA, there are no changes in configuration and operation except in the following areas:

- For static pad setting, the configuration for short line and long line has been changed from TRC to SHL for short line, and NTC to LOL for long line.
- The PPM configuration is done on a per route basis.
- Only one value is now downloaded for the PPM ID, on all UK cards.
- Only COT trunks are supported on the NT5K18AA. The NTAG04AA card supports COT and DID trunks.
- The balance impedance may now be configured on the NT5K90AA (Danish PPM) or NT5K90BA (Danish non-PPM) card.

Feature packaging

Meridian 1 Superloop Administration (XCT1) package 205.

Dependencies: Intelligent Peripheral Equipment (XPE) package 203; PPM/ Message Registration (MR) package 101; Trunk Failure Monitor (TFM) package 182; and Trunk Hook Flash (Centrex) (THF) package 157.

Feature implementation

LD 14 – Configure the trunk parameters for the new XFCOT cards.

Prompt	Response	Description
...		
CDEN	4D 8D	Card Density, where: 4D = Quad density, and 8D = Octal density.
...		
SIGL	ALS	ALS signaling on COT trunk with ground start (applies to the NTAG04AA unit only).
BIMP	(3COM) 600	Three-component complex impedance. 600 ohms.
...		
SEIZ	(NO) YES	Automatic Guard Detection for outgoing trunk.

PPID	(0)-15	PPM country ID. Must be configured if PPM is enabled on the route. One PPID type per card. Trunks must be removed from a card to change PPID. Choose from one of the following PPM IDs, according to country: (0) – UK (50 Hz). 1 – France (50 Hz). 2 – France (12 Hz). 3 – Germany (16 kHz). 4 – Switzerland (12 kHz). 5 – Denmark (12 kHz). 6 – Norway (16 kHz). 7 – Belgium (16 kHz). 8 – Spain (12 kHz). 9 – Portugal (12 kHz). 10 – Holland (50 Hz). 11-15 – Reserved for future use.
BTID	(0)-15	Busy Tone Country ID. Must be configured for BTS supervised XCOT trunk. One BTID type per card. Trunks must be removed from card to change BTID. Choose from one of the following Busy Tone IDs, according to country: (0) – CCITT. 1-2 – Reserved fro future use. 3 – Germany. 4 – Switzerland. 5 – Denmark. 6 – Norway. 7-9 – Reserved for future use. 10 – Holland. 11-15 – Reserved for future use.

CLS		Class of Service options.
	(SHL) LOL	Enter SHL for short line (LOL for long line) static loss pad selection.
	(XBAT) BAT	Enter BAT for battery supervised COT; (XBAT) for no battery supervision.
	(XARF) ARF	Enter ARF for ARF supervised COT; (XARF) for no ARF supervision.
	(XLBS) LBS	Enter LBS for loop break supervised COT; (XLBS) for no loop break supervision.
	(XBTS) BTS	Enter BTS for tone supervised COT; (XBTS) for no tone supervision.
	(BARD) BARA	Enter BARA to allow barring; (BARD) to deny barring.
SUPN	(NO) YES	Enter SUPN = NO or the appropriate supervision type.
STVP		Entering any of the following prompts will now override the previously configured type.
	BAT	Enter BAT for battery supervised COT.
	ARF	Enter ARF for ARF supervised COT.
	LBS	Enter LBS for loop break supervised COT.
	BTS	Enter BTS for tone supervised COT.
		Note: The XBAT, XARF, XLBS, and XBTS prompts are no longer applicable.
Note: SHL/LOL, BARA/BARD remain appropriate responses for the CLS prompt.		

Feature operation

No specific operating procedures are required to use this feature.

Extended Multifrequency Compelled Sender/Receiver

Contents

This section contains information on the following topics:

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Feature implementation 308
Feature operation 309

Feature description

This feature provides combined Multifrequency Compelled Signaling (MFC) and MFE signaling for SOCOTEL, using the Extended Multifrequency Compelled Sender/Receiver NT5K21AA card. This card is based on the XDTR card NT8D16AB.

Although the NT5K21AA card provides both MFC and MFE signaling, it may only be configured as one or the other: it is not possible to configure certain units as MFC and other units as MFE, on the same card. If there is a requirement for both MFC and MFE signaling, then two NT5K21AA cards may be configured – one for MFC and one for MFE.

In support of the NT5K21AA card, the system software has been modified as follows:

- Four DS-30X channels are provided for simultaneous generation and detection (forward and backwards) of MFC digits
- Four DS-30X channels are provided for alternate generation and detection of MFE digits (software selectable)
- A DS-30X channel of A10 formatted signaling is provided for communication between the system CPU and the NT5K21AA pack
- A-law and μ-law PCM encoding schemes are both supported
- Any one of 16 tone output levels may be specified for each channel
- Any one of four levels may be specified as a minimum receiver acceptance level
- Special MFC functions, such as pulse or automatic mode, are provided
- Card-ID information, configured during the manufacture of the NT5K21AA pack, is stored in the EEPROM message
- Hardware self-test and troubleshooting capabilities, including loop-back of PCM channels at the NT5K21AA, are provided by maintenance software, and
- The standard faceplate Enable/Disable Status Indicator LED is provided.

Most of the existing command structure for signaling has been maintained, with the following exceptions:

- During RESET, the NT5K21AA card is configured as either MFC or MFE, and as either A-law or μ-law
- More comprehensive self-test results are provided
- The minimum receiver acceptance level (MFL) is downloaded, and
- An extended range is provided for the MFC digit level (MFL).

Operating parameters

Both A-law and µ-law, which are software selectable, are supported. But when a Companding Law is selected in LD 97, it is supported on a system basis.

System parameters have to be downloaded on the NT5K21AA card in the following cases:

- When the NT5K21AA card is enabled in LD 32 and 54
- During service changes and initialization
- When a new NT5K21AA unit is defined in LD 13, and
- When an NT5K21AA card is moved to another card, in LD 13.

The default system parameters for downloading are NT5K21AA card type MFC, Mµ-law companding law, and a Minimum Receiver Acceptance level of -36 dB.

The following Card-LAN interface capabilities are supported by the NT5K21AA card:

- Periodic Intelligent Peripheral Equipment (IPE) polling of the status of the NT5K21AA card
- Requesting of card-ID, card type, and firmware version for auto-configuration, and
- Requesting of configuration data, including the DS-30X signaling type, during power up and RESET.

The auto-configuration of the NT5K21AA card is not supported on Small Systems and Succession 1000 systems.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

- Intelligent Peripheral Equipment (XPE) package 203.

- Multifrequency Compelled Signaling (MFC) package 128.
- Multifrequency Signaling for SOCOTEL (MFE) package 135.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 97 – Download the system parameters, and define the Multifrequency Minimum Receiver Level (MFRL).

2 LD 94 – Create the MFC/MFE Signaling Level tables.

3 LD 13 – Create the MFC/MFE unit data block.

4 LD 16 – Create the route data block.

5 LD 14 – Create the trunk data block, and define the range of Multifrequency Digit Level (MFL).

LD 97 – Download the system parameters, and define the Multifrequency Minimum Receiver Level (MFRL).

Prompt	Response	Description
...		
MFRL	0-(2)-3	Multifrequency minimum Receiver Level for XMFC/XMFE (NT5K21) for Superloop only. 0 = -28 dBM. 1 = -32 dBM. 2 = -36 dBM (the default). 3 = -40 dBM.

LD 94 – Create the MFC/MFE Signaling Level tables.

LD 13 – Create the MFC/MFE unit data block.

LD 16 – Create the route data block.

LD 14 – Create the trunk data block, and define the range of Multifrequency Digit Level (MFL).

Prompt	Response	Description
...		
MFL	(0)-15	Multifrequency digit level. Expanded from 0-7 to 0-15 for Superloop only. Enter the MFC digit level required for signals to the Public Switched Telephone Network (PSTN). Superloop codes and values: 0 = -8 dBM. 1 = -11 dBM. 2 = -12 dBM. 3 = -13 dBM. 4 = -14 dBM. 5 = -15 dBM. 6 = -16 dBM. 7 = -31 dBM. 8 = -4 dBM. 9 = -5 dBM. 10 = -6 dBM. 11 = -7 dBM. 12 = -9 dBM. 13 = -10 dBM. 14 = spare. 15 = spare. ***Note:*** Levels 0-7 are already defined.

Feature operation

No specific operating procedures are required to use this feature.

Extended Tone Detector Global Parameters Download

Contents

This section contains information on the following topics:

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Feature packaging 313
Feature implementation 313
Feature operation 316

Feature description

An Extended Tone Detector (XTD) card is capable of performing both Dual-tone Multifrequency (DTMF) and Dialtone (DT) detection. It is possible to download parameters onto the card so that it may be customized for a particular environment. On the current UK Extended Tone Detector (XTD) cards, the NT5K20AA and the NT5K20AB, it is possible to download two parameters onto these cards. These parameters are the A-law/Mμ-law for the Extended DTMF (XDTR) portion of the card, and the first stage dialtone detection (DT).

This feature allows several new parameters and a new message to be downloaded onto the new global XTD pack, the NT5K48AA. The new parameters, grouped under the categories of first stage dialtone detection, second stage dialtone detection, and XDTR minimum accept level, are:

- flexible first stage dialtone detection
  - frequency band (expanded operation)
  - minimum detect level
  - minimum validation time
  - break duration
  - cadence type
- flexible second stage dialtone detection
  - second stage configuration, and
- flexible XDTR minimum accept level

The new message is the Detect Second Stage Dialtone. It allows the NT5K48AA to distinguish between using the first stage dialtone detection parameters and the second stage dialtone detect parameters for detecting dialtone.

To configure the first and second stage dialtone detection parameters, a new type, DTD, and associated prompts have been introduced in LD 97. This prompt allows a craftsperson to create up to eight different XTD tables containing the parameters. In LD 13, a table is associated with each XTD card. These parameters are downloaded onto each XTD card.

To configure the flexible XDTR minimum accept level parameter, a new type, DTR, and associated prompt MINL (that defines the minimum accept level, on a per-system basis) have been introduced in LD 97. This parameter is downloaded onto each XTD card and DTR card.

Operating parameters

The global NT5K48AA card supports the first stage dialtone detection of the NT5K20AA and the NT5K20AB cards. Although the NT5K20AA and the NT5K20AB can be used with the NT5K48AA, these UK cards do not support second-stage dialtone detection because they cannot interpret the new Detect Second Stage Dialtone message (second stage dialtone detection is not used for the UK market).

Since there is only one parameter for the second stage dialtone detector (the craftsperson, in LD 97, enters a value between 0-15 to indicate which of the 16 options to use), the parameters for second stage dialtone detection hardware operation are hardcoded with limited flexibility. The NT5K48AA has to be modified to provide second stage configuration, if it is to be introduced to a country that has an undefined configuration.

The default values for all parameters are for the Swiss standards. However, if the UK Program (UK) package 190 is equipped, the UK recommended default values are used.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Intelligent Peripheral Equipment (XPE) package 203; and Meridian 1 Superloop Administration (XCT1) package 205.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 97 – Configure all the first- and second- stage dial tone detection parameters (TYPE = DTD).

2 LD 97 – Configure the flexible XDTR minimum accept level parameter (TYPE = DTR).

3 LD 13 – Define the protected data block of the XTD card.

4 LD 16 – Define the route protected data block of the XTD card.

LD 97 – Configure all the first- and second- stage dial tone detection parameters (TYPE = DTD).

Prompt	Response	Description
...		
TYPE	DTD	First- and second-stage dialtone detection parameters.
XTDT	(0)-7	Extended Tone Detection Table. XDTD table number in which the parameters are stored. Table 0 can be changed but must not be removed. Table 0 always exists and is initialized to default values.
DFQ	0-(4)-15	Dial Tone Frequency band for 1st dial tone, which is the number of the dial tone frequency band chosen in the hardcoded frequency table. With United Kingdom (UK) package 190 the default value for DFQ = 0.
MDL	10-(20)-40	Minimum Detect Level for 1st dial tone in dBm, which is the absolute value of the minimum detect level. Odd input is rounded down. With United Kingdom (UK) package 190 the default value for MDL = 30 (-30 dBM).
MVT	100-(400)-1600	Minimum Validation Time for dial tone in milliseconds. Input that is not a multiple of 100 is rounded down to the next multiple of 100. With United Kingdom (KUK) package 190 the default value for MVT = 300.
BRK	(0)-240	Break Duration (maximum) for 1st dial tone in milliseconds. Input that is not a multiple of 16 is rounded down to the next multiple of 16.
CAD	(0)-15	Cadence type for 1st dial tone, which is the number of the cadence pattern in the hardcoded table.
SSC	(0)-15	Second Stage Configuration, which is the configuration number for the second stage dial tone detection to be set in the firmware.

LD 97 – Configure the flexible XDTR minimum accept level parameter (TYPE = DTR).

Prompt	Response	Description
...		
TYPE	DTR	First- and second-stage dial tone detection parameters.
MINL	3-(42)-48	Minimum accept level for Digitone Receivers in dBM, which is the absolute value of the minimum accept level. Input that is not a multiple of 3 is rounded down to a valid multiple of 3. With United Kingdom (UK) package 190 the default value for MINL = 45 (-45 dBm).

Refer to Table 6 for recommended configuration values for each country. The default values given in parenthesis are for non-UK countries.

Table 6
Recommended parameters according to country (Part 1 of 2)

Country	DFQ	MDL	MVT	BRK	CAD	SSC	MINL
Germany	1	-16 dBm	1000 ms.	0 ms.	0	—	-45 dBm
France	0	-24 dBm	1000 ms.	30 ms.	0	0	-30 dBm
Sweden	1	-28 dBm	1000 ms.	60 ms.	0	—	-28 dBm
Norway	1	-32 dBm	1400 ms.	0 ms.	0	—	-45 dBm
Switzerland	4	-28 dBm	1000 ms.	0 ms.	0	—	-30 dBm
Spain	2	-32 dBm	1000 ms.	0 ms.	0	0	-30 dBm
UK (330/440)	0	-30 dBm	500 ms.	0 ms.	0	—	-45 dBm
UK (33/50)	3	-30 dBm	900 ms.	0 ms.	0	—	-45 dBm

Table 6
Recommended parameters according to country (Part 2 of 2)

Country	DFQ	MDL	MVT	BRK	CAD	SSC	MINL
Denmark	1	TBD	TBD	0 ms.	0	—	-45 dBm
Holland	0	TBD	TBD	TBD	0	—	-30 dBm
New Zealand	1	TBD	TBD	TBD	0	—	-45 dBm

LD 13 – Define the protected data block of the XTD card.

Prompt	Response	Description
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted when TYPE = XTD. If a table other than 0 is entered, it must have already been configured in LD 97.

LD 16 – Define the route protected data block of the XTD card.

Prompt	Response	Description
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted with Meridian 1 Superloop Administration (XCT1) package 205. Must be the same value as defined in LD 13. If a table other than 0 is entered, it must have already been configured in LD 97.

Feature operation

No specific operating procedures are required to use this feature.

Fast Tone Digit Switch

Contents

This section contains information on the following topics:

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Feature operation 318

Feature description

The QPC609 Fast Tone and Digit Switch (FTDS) card, along with the associated software, can reduce call setup time by as much as 50 percent with features such as Basic/Network Alternate Route Selection (BARS/NARS), Stored Number Redial, Speed Call, and System Speed Call. With the use of an on-board buffer memory, the calling efficiency of end users is greatly improved.

The QPC609 can be operated in two different modes as defined by the customer: either with 100 milliseconds (ms) Dual-tone Multifrequency (DTMF) bursts or with 50 ms DTMF bursts. The software can load up to 32 digits into the buffer in a single time slice, and can outpulse the digits at a maximum rate of 10 digits per second.

Operating parameters

Tone Digit Switch cards QPC197 and QPC251 cannot coexist with the QPC609 or NT8D17 within the same system.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Fast Tone and Digit Switch (FTDS) package 87 has no feature package dependencies.

Feature implementation

LD 17 – Change the duration of Digitone burst.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System parameters.
PARM	(NO) YES	(Do not) change system parameters.
- DTRB	50 60 70 (100)	Digitone burst time in milliseconds.

Feature operation

No specific operating procedures are required to use this feature.

FCC Compliance for DID Answer Supervision

Contents

This section contains information on the following topics:

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Feature interactions 324
Feature packaging 325
Feature implementation 325
Feature operation 325

Feature description

This feature is designed to meet the requirements in the United States, Section 68.314(h) of Part 68, and the DOC requirements in Canada, Section 3.22 of CSO3 Part 1, for answer supervision of redirected telephone calls to ensure proper billing.

This feature is designed specifically for telephone calls coming in through Direct Inward Dialing (DID) trunks. Answer supervision for all other types of telephone calls is not affected. This feature works in conjunction with the following types of calls:

- Direct Inward Dialing (DID) calls terminating at the system and forwarded to a Recorded Announcement (RAN).
- Direct Inward Dialing (DID) calls forwarded by the system through the public switched network (PSN) to another number in the Public Exchange/Central Office (CO), or to another system.

  On North American COT, FEX, and WATS trunks, Central Offices do not always return answer supervision. When no answer supervision is returned, the system software uses the end-of-dial timer for non-Digitone trunks (EOD timer), or the end-of-dial timer for Digitone trunks (ODT timer) to verify call connection. For Federal Communications Commission (FCC) compliance, the EOD and ODT timers will still be used for incoming DID calls, except that EOD is capped at 20 seconds even if configured for more.

This feature handles incoming DID calls over Data Terminal Interface (DTI), Integrated Services Digital Network (ISDN), and analog trunks. Outgoing calls over Central Office (CO) and TIE trunks are also handled. System components involved include trunks, the system, and the CO.

The following explains how the system components handle answer supervision.

- Analog, DTI, and ISDN incoming trunks: These are covered as long as they are DID incoming trunks. For incoming analog and DTI trunks, answer supervision or pseudo-answer supervision is returned by the system to the CO, if necessary. For incoming ISDN trunks, the connect message is returned instead.
- Analog, DTI, and ISDN outgoing trunks: For incoming DID calls, the answer and disconnect supervisor (SUPN) of the outgoing trunk is forced to NO. The EOD or ODT timer simulates the return of answer supervision.

- For DID calls terminating at the system, the system returns answer supervision based on the terminating condition. For DID calls forwarded to Public Switched Networks (PSN) or private networks, the system returns answer supervision based on the condition of the outgoing trunk (whether answered or timed out).
- CO: The system provides the pseudo-answer for DID calls because the system cannot return answer supervision.

DID calls terminating at the system

The requirements for a DID call terminating at the system to return answer supervision to the incoming DID trunk are shown in the following table. The ASUP prompt in LD 16 is kept for other types of calls, but the system software enforces the correct settings to return answer supervision if a Recorded Announcement (RAN) is used for DID calls, regardless of the value originally specified in the service change.

Table 7
Returning Answer Supervision for DID calls terminating at the system

DID call terminating status	Answer supervision returned with FCC Compliance
Answered by the called DID station	Yes
Answered by an attendant	Yes
Routed to dialing prompt	Yes
Routed to Meridian Mail	Yes
Routed to Recorded Announcement, including invalid number, not in service, and not assigned announcements	Yes
Routed to Recorded Announcement by Automatic Call Distribution (ACD), including invalid number, not in service, and not assigned announcements	Yes

DID call terminating status	Answer supervision returned with FCC Compliance
Not answered	No
Busy signal	No
Recorder signal	No

Calls forwarded to Public Switched Network

Because it is uncertain whether or not the far end will return answer supervision, the system uses the EOD and ODT timers. If the system has not detected the return of answer supervision upon timeout of the outgoing CO trunk, the system sends pseudo-answer supervision to the incoming DID trunk. This timer is set in LD 16 on a per-route basis. When a CO trunk is configured, system software forces the value of SUPN to NO. Consequently, system software does not expect the return of answer supervision, and returns answer supervision in the following cases:

- The system receives answer supervision from the outgoing CO trunk before the EOD or ODT timer of the outgoing route expires.
- The system does not receive answer supervision from the outgoing trunk and the EOD or ODT timer of the outgoing route expires; pseudo-answer supervision is generated.

  Note: There are still some cases in which the SUPN value for CO trunks is assigned to YES if the CO supports a reverse battery mechanism.

With FCC Compliance, a more stringent mechanism is introduced to apply SUPN = NO in LD 14 to all CO trunks, even those configured as polarity sensitive. Service-changeable EOD or ODT timers are always used for incoming DID calls to enforce the return of answer supervision. In this case:

EOD = 128-19,968 milliseconds (ms) (default time is 13,952 ms), and

ODT = 256-16,128 ms (default time is 4,096 ms).

The EOD timer expires at 20 (20,000 ms) for FCC Compliance. For outgoing DID calls, the EOD upper limit is 32,640 ms.

DID calls forwarded to private networks

Answer supervision is not always returned on TIE trunks because some TIE trunks leased from public carriers are connected to COs that do not support answer supervision.

Currently, the system provides the SUPN prompt (LD 14) to specify the availability of answer supervision on certain types of trunks, including TIE, CAM, Common Control Switching Arrangement (CCSA), and CAA (CCSA Automatic Number Identification [ANI]). If SUPN is YES, and it is an outgoing trunk, the system does not return answer supervision to the incoming DID trunk unless answer supervision is received from that outgoing trunk. If the user specifies NO, the system returns pseudo-answer supervision upon EOD or ODT timeout. Such implementation causes short billing and overcharge problems.

To solve this problem, a treatment similar to the one implemented on CO trunks is used on the trunks in this category. The system enforces SUPN = NO without changing the SUPN value.

For incoming DID calls routed to private networks, SUPN is enforced to NO to ensure the return of answer supervision on the outgoing TIE, CO, FEX, WATS, CAM, CAA, and CCSA trunks. If answer supervision is not returned when the end of dial timeout occurs, the system disregards the original value of SUPN set by the user and forces the return of answer supervision.

When the call comes from a DID trunk, the following outgoing trunks are affected: TIE, CO, FEX, WATS, CAM, CAA, and CCSA.

Operating parameters

Allowing system equipment to be operated in such a manner as to not provide proper answer supervision signaling is in violation of Part 68 rules.

This equipment returns answer supervision signals to the public switched telephone network (PSTN) when:

- answered by the called station
- answered by the attendant

- routed to a Recorded Announcement that can be administered by the Customer Premises Equipment (CPE) user, and
- routed to a dial prompt.

This equipment returns answer supervision on all DID calls forwarded back to the PSTN. Permissible exceptions are when:

- a call is unanswered
- a busy tone is received, and
- a reorder tone is received.

Feature interactions

Extended DID/DOD Software Support - Europe

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XDOD units, it may lead to incorrect call status. Therefore, equipping this package is not recommended.

Extended Flexible Central Office Trunk Software Support

If FCC Compliance for DID Answer Supervision (FC68) package 223 is configured on XFCOT units, it may lead to incorrect call status. Therefore, equipping the FCC package is not recommended.

Feature Group D and Japan DID trunks

Feature Group D trunks and Japan (JPN) DID trunks are not affected by this feature.

ISDN trunks

Both incoming and outgoing Integrated Services Digital Network (ISDN) trunks are affected by this feature.

- For ISDN incoming DID trunks, the connect message is returned when answer supervision is returned or when the end of dial timer expires.
- For ISDN outgoing trunks, the end of dial timer is added to the protocol to simulate the EOD timer when a connect message is not returned from the far end; the system generates a pseudo-answer supervision to send to the incoming trunk.

Intercept
Recorded Announcement

With this feature, incoming DID calls that are intercepted to a Recorded Announcement (RAN) are provided with answer supervision.

Feature packaging

This feature requires Federal Communications Commission Compliance for DID Answer Supervision (FC68) package 223.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

FCC Compliance for Equal Access

Contents

This section contains information on the following topics:

Feature description 327
Operating parameters 329
Feature interactions 330
Feature packaging 330
Feature implementation 330
Feature operation 332

Feature description

This feature brings the system into compliance with the Equal Access portion of the Federal Communication Commission (FCC) 68 ruling. This calls for the optional restriction of two types of direct-dialed Equal Access toll calls, while allowing all other Equal Access dialing sequences (with the exception of operator cut-through) and call processing operations.

The two types of Equal Access calls that may be restricted are:

- North American toll calls (1+NPA+NXX+XXXX),where NPA = Number Plan Area, NXX = any three digits with N being any digit except 0 or 1, and XXXX = any four digits, and
- International toll calls (011+CC+NN),where CC = Country Code and NN = National Number.
  FCC compliant dialing plans.

Table 8
FCC compliant dialing plans.

Dialing Format	Destination
Allow:	
10XXX+0	Operator of carrier specified by XXX.
10XXX+0+(NPA)+NXX+XXXX	Operator function of carrier specified by XXX.
10XXX+0+SAC+NXX+XXX	Subscribed carrier specified by XXX.
10XXX+01+CC+NN	Operator function of carrier specified by XXX.
Allow/Deny:	
10XXX+1+(NPA)+NXX+XXXX	Carrier specified by XXX.
10XXX+011+CC+NN	Carrier specified by XXX.
where: XXX = any three digits, XXXX = any four digits, NPA = Number Plan Area, NXX = any three digits with N being any digit except 0 or 1, CC = Country Code, NN = National Number.	

This feature provides two methods of restricting Equal Access toll calls, General Carrier Restriction (GCR), and Selective Carrier Restriction (SCR). These restrictions, configured in LD 16, require that the originating set have a Network Class of Service of Equal Access. The Equal Access restriction for an NCOS group is configured in LD 87.

GCR permits a configuration of allowing or denying all North American Equal Access toll calls and all international Equal Access toll calls. This GCR restriction is based on call type only, and does not take into account the dialed Carrier Identification Code. SCR uses the New Flexible Code Restriction (NFCR) feature to place a more selective restriction on Equal Access toll calls, based on the dialed Carrier Identification Code (CIC). So, for example, Equal Access toll calls for a carrier with a CIC of 434 could be denied, while Equal Access toll calls for a carrier with a CIC of 225 could be allowed.

GCR is the simplest method to implement and requires no additional memory. It is therefore recommended that GCR be used if there is no need to restrict Equal Access toll calls based on carrier usage. SCR is more difficult to set up and requires additional memory. Use this method only if there is a strong need to restrict Equal Access toll calls based on carrier usage.

Since both methods can be active at the same time, the optimum solution in some cases would be to implement a combination of GCR and SCR. If, for example, a requirement exists to restrict all North American Equal Access toll calls and only certain international Equal Access toll calls, based on carrier usage, then GCR could be configured to handle the North American Equal Access toll calls while SRC could be configured to handle the international Equal Access toll calls.

Operating parameters

The same requirements for normal calls using the New Flexible Code Restriction (NFCR) feature apply to calls made under the Selective Carrier Restriction method, except that Equal Access operator calls (10XXX0) are allowed to be completed while Equal Access international toll calls (10XXX011) are denied.

This feature could require extra memory when operating under the Selective Carrier Restriction method (as much as 15.5K words of protected data storage when fully configured). Insufficient memory may limit the number of CIC codes which may be restricted.

This feature only supports COT, FEX, WAT, DID, and TIE routes with Standard Signaling.

This feature does not support network signaling, since the intention is to restrict Equal Access calls directly terminating at the Central Office and not at another network node.

This feature does not restrict calls made by an attendant.

The # sign is not outpulsed by the system, as recommended in the FCC Bellcore North American Dialing Plan.

The operator cut-through dialing sequence of 10XXX#, which is recommended in the FCC Bellcore North American Dialing Plan, is not supported on the system.

Feature interactions

New Flexible Code Restriction

The New Flexible Code Restriction (NFCR) feature has been modified to allow for the restriction of Equal Access international toll calls (10XXX+011+CC+NN) while not restricting Equal Access operator calls (10XXX+0).

Feature packaging

This feature is not packaged, however the following packages are required to make it operational: Network Class of Service (NCOS) package 32 is required for both the General Carrier Restriction and Selective Carrier Restriction methods; and New Flexible Code Restriction (NFCR) package 49 is required for the Selective Carrier Restriction method.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 16 – Apply Equal Access call restriction to this route.
2. LD 87 – Specify whether Equal Access with a NCOS group is to be associated or not.

LD 16 – Apply Equal Access call restriction to this route.

Prompt	Response	Description
...		
EQAR	(NO) YES	Enable Equal Access Restrictions. Prompted when TKTP = CO, FEX, WAT, or ISA, and ICOG = OGT, or IAO.
- GCR	(NO) YES	General Carrier Restriction to restrict Equal Access calls.
- - NTOL	(DENY) ALOW	North American toll calls (that is, 1+ calls).
- - ITOL	(DENY) ALOW	International toll calls (that is, 011+ calls).
- SCR	(NO) YES	Selective Carrier Restriction to restrict Equal Access calls. Prompted when EQAR = YES, and New Flexible Code Restriction is enabled. NTOL and ITOL must both be ALOW.

LD 87 – Specify whether Equal Access with a NCOS group is to be associated or not.

Prompt	Response	Description
...		
- EQA	(NO) YES	Equal Access (is not) is associated with this NCOS group.

Feature operation

The dialing sequence for Equal Access calls on the system is:

- Access Code (either trunk or NARS/BARS)
- Carrier Access Code (CAC). The CAC is comprised of the Equal Access code (10) and the Carrier Identification Code (CIC) (any three digits). The CIC specifies the carrier that will handle the call,
- Telephone number.

The dialing sequence can contain two special characters, the asterisk (*) and the number sign (#). The * sign within a dialing invokes a three-second pause in the call processing procedure, and has no bearing on call restriction routines. The # sign within a dialing sequence signifies the end of the dialing sequence, and that it can be examined by call restriction routines. The only exception occurs when all international Equal Access toll calls have been restricted on a switch. In this case, direct-dialed Equal Access operator calls may not terminate with the # sign (in order to avoid possible fraud when calls are placed from trunks with Digitone Class of Service).

First-second Degree Busy Indication

Contents

This section contains information on the following topics:

Feature description 333
Operating parameters 333
Feature interactions 333
Feature packaging 334
Feature implementation 334
Feature operation 334

Feature description

This feature provides an attendant with an indication of whether a party is first degree or second degree busy. If party A is established on a call to party B, and the attendant tries to connect to party A, party A is considered to be first degree busy. If party C is camped-on or call waiting to party A, party A is then considered to be second degree busy.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged in International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – At the OPT prompt, deny/allow Attendant Busy Display.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	ATT	Attendant Console Options
...		
- OPT	(ABDD) ABDA	Attendant Busy Display (denied) allowed.

Feature operation

The first degree busy indication is as normal. For second degree busy indication, normal busy tone is given to the attendant, and the display -O (meaning Occupied Second Degree) is given on the last four right-hand spaces of the console display.

Flexible Attendant Call Waiting Thresholds

Contents

This section contains information on the following topics:

Feature description 335
Operating parameters 336
Feature interactions 336
Feature packaging 336
Feature implementation 337
Feature operation 339

Feature description

When there are no calls waiting in the attendant queue, the Call Waiting Lamp on all Attendant Consoles is dark. The lamp is lit as soon as the first call arrives that can not be presented to a console.

When the number of calls waiting in the attendant queue exceeds the upper threshold, defined by the CWCL prompt in LD 15, the Call Waiting Lamp (CWL) state on all Attendant Consoles is changed from lit to flash (60 impulses per minute).

When the number of calls waiting in the attendant queue drops below the lower threshold, defined by the CWCL prompt in LD 15, the CWL state on all Attendant Consoles is changed from flash to lit.

When there are no more calls waiting in the attendant queue, the CWL is turned off.

The Flexible Attendant Call Waiting Thresholds (FACWT) feature allows the thresholds to be defined as a percentage of the active consoles, consoles which are not in Position Busy or Night Service, or as a fixed number. The feature is activated on a customer basis by responding with FACA (Flexible Attendant Call Waiting Thresholds Allowed) to the OPT (Option) prompt in LD 15.

Operating parameters

The upper threshold must be greater than or equal to the lower threshold.

The maximum number of attendants multiplied by the threshold maximum percentage must equal less than 65 535 (due to storage requirements).

Feature interactions

Attendant Overflow Position

The Attendant Overflow Position is not counted as an active attendant.

Recall to Same Attendant

The Recall to Same Attendant (RTSA) feature has precedence over the Flexible Attendant Call Waiting Thresholds feature. If either RSAA or RSXA options are selected, RTSA has precedence over FACWT in determining the Call Waiting Lamp state. If one or more RTSA calls are waiting in the attendant queue, RTSA will set the Call Waiting Lamp state to wink (30 impulses per minute).

RTSA calls are not included when the FACWT feature determines the number of calls waiting.

Feature packaging

The Flexible Attendant Call Waiting Thresholds is packaged under International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – The Customer Data Block service change accepts the options FACD and FACA to be defined as a customer option. The range and usage of the CWCL thresholds is defined by the FAC option selected. To allow the calls waiting thresholds to be defined as percentages respond to the OPT prompt with FACA. To allow the calls waiting thresholds to be defined as number of calls respond to the OPT prompt with FACD.
2. LD 21 – Print Routine 2 is modified to include OPT FACD or FACA setting and the new CWCL range settings in the Customer Data Block printout.
3. LD 93 – As for the Customer Data Block, the CWCL threshold usage is changed with the selection of a FAC option in the Customer Data Block.

LD 15 – The Customer Data Block service change accepts the options FACD and FACA to be defined as a customer option. The range and usage of the CWCL thresholds is defined by the FAC option selected. To allow the calls waiting thresholds to be defined as percentages respond to the OPT prompt with FACA. To allow the calls waiting thresholds to be defined as number of calls respond to the OPT prompt with FACD.

Note: If OPT is changed from FACD to FACA, or from FACA to FACD, a new value must be set for CWCL in LD 15 or the default values (0 0) will be used. The values of the call waiting thresholds for the tenant level in LD 93 are set equal to the customer level LD 15 values. A service message is output when the values are set.

Prompt	Response	Description
REQ:	CHG NEW	Modify or create data block.
TYPE:	ATT	Attendant Console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems

...		
- OPT	(FACD) FACA	Options for customer: (Flexible Attendant Call Waiting Thresholds Denied), Flexible Attendant Call Waiting Thresholds Allowed. (Denies), Allows the attendant Call Waiting thresholds to be defined as a percentage of active attendants.
...		
- CWCL	xxxyyy	Call Waiting Lamp thresholds Where xxx defines the lower threshold and yyy defines the upper threshold.
	(0)-255(0)-255	Valid ranges for number of calls when FACD is entered in response to OPT.
	(0)-1000(0)-1000	Valid ranges for percentages when FACA is entered set in response to OPT.

LD 21 – Print Routine 2 is modified to include OPT FACD or FACA setting and the new CWCL range settings in the Customer Data Block printout.

Prompt	Response	Description
REQ	PRT	Request: Print data block.
TYPE	CDB	Customer Data Block
CUST	xx	Customer number, as defined in LD 15

LD 93 – As for the Customer Data Block, the CWCL threshold usage is changed with the selection of a FAC option in the Customer Data Block.

Note: The lower and upper call waiting thresholds must be redefined whenever they are changed between number of calls and percentage definition. Respond to the CWCL prompt with the new definitions.

Prompt	Response	Description
REQ	CHG NEW PRT	Request: Modify, create or print data block.
TYPE	CPGP	Console Presentation Group Parameters.

CUST	xx	Customer number, as defined in LD 15
CPG	1-63	Console Presentation Group: ACG (Attendant Console Group) number.
...		
AODN	...	
CWCL	xxxyyy	Call Waiting Lamp thresholds Where xxx defines the lower threshold and yyy defines the upper threshold.
	(0)-255(0)-255	Valid ranges for number of calls when FACD is set in response to OPT in LD 15.
	(0)-1000(0)-1000	Valid ranges for percentages when FACA is set in response to OPT in LD 15.

Feature operation

If the customer has the FACA option selected in the Customer Data Block (LD 15) the thresholds are defined as a percentage of the number of active attendants. The thresholds are specified on a customer and tenant Console Presentation Group (CPG) level basis. If the Flexible Attendant Call Waiting Thresholds Denied (FACD) option is selected, the thresholds are defined as fixed numbers and the operations remain the same as when this feature is not used.

When the FACA option is used, the CWL state is updated each time the number of calls waiting or the number of active attendants changes. Any integer between 0-1000 can be set for the Call Waiting thresholds percentage. The following tables illustrate the operation when FACA is selected and the lower limit is defined as 100 percent of active attendants and the upper limit is defined as 200 percent of active attendants (CWCL 100 200).

Table 9
Upper and lower limits of calls waiting versus number of active attendants

Number of active attendants	Number of calls waiting in queue to achieve 100% lower limit	Number of calls waiting in queue to achieve 200% upper limit
1	1	2
2	2	4
3	3	6

Table 10
CWL state versus number of active attendants

Number of active attendants	Number of calls in queue												
	0	1	2	4	6	8	6	4	3	2	1	0	
1	D	L	L	F	F	F	F	F	F	F	L	D	CWL state
2	D	L	L	L	F	F	F	F	F	L	L	D	
3	D	L	L	L	L	F	F	F	L	L	L	L	

Legend: D = Dark, L = Lit, F = Flash.

Flexible Attendant Directory Number

Contents

This section contains information on the following topics:

Feature description 341
Operating parameters 341
Feature interactions 342
Feature packaging 342
Feature implementation 342
Feature operation 342

Feature description

The Flexible Attendant Directory Number (FADN) specifies the Directory Number (DN) that provides access to the attendant, replacing the usual 0. The DN may be any DN in the numbering plan, but it must be used only for the attendant and in all situations in which 0 is normally used.

Operating parameters

The attendant DN may be used only for the attendant. One attendant DN is allowed per customer and all attendants must have the same DN.

Feature interactions

Directory Number Expansion

The attendant DN can have up to seven digits if the Directory Number Expansion (DNXP) package is equipped.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Define or change the attendant Directory Number.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Attendant console options
...		
- ATDN	xxx...x	Number dialed to reach the attendant (the default is 0).

Feature operation

No specific operating procedures are required to use this feature.

Flexible Busy Tone Timer

Contents

This section contains information on the following topics:

Feature description 343
Operating parameters 343
Feature interactions 343
Feature packaging 344
Feature implementation 344
Feature operation 344

Feature description

The feature provides a flexible length of time that a caller on a Direct Inward Dialing (DID) route hears busy or overflow tone, when it is normally encountered. The time that the tone is presented is overlay programmable from 2 to 254 seconds.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged in the International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Set data for Flexible busy/overflow time to implement the flexible Busy Tone Timer feature:

Prompt	Response	Description
...		
BTT	2-(30)-254	Enter busy/overflow time to be returned on DID routes in seconds.

Feature operation

No specific operating procedures are required to use this feature.

Flexible Dial Tone Detection

Contents

This section contains information on the following topics:

Feature description 345
Operating parameters 346
Feature interactions 346
Feature packaging 346
Feature implementation 347
Feature operation 347

Feature description

The Flexible Dial Tone Detection (FDTD) feature permits the system to wait for and detect a Second Dial Tone (SCDT) before automatic or manual dialing of outgoing toll calls. The wait-for-tone position in the digit outpulsing is user configurable, thus providing flexible digit validation. This feature is an enhancement to the Dial Tone Detection (DTD) feature.

The break-in outpulsing can occur after a defined digit sequence, or after a defined number of digits have been outpulsed. Digit outpulsing is halted and the Dial Tone Detector is reconnected. With the FDTD feature, it is no longer necessary to use the * to create pauses in outpulsing.

This feature has the following three options:

Dial Tone Position (DTP)

With the DTP option an Outgoing Access Code (OAC) is selected. Then FDTD verifies the dialed digits against the OAC (for example, country code) of up to four digits. When DTP is set, only the OAC digits are outpulsed before the DTD is reconnected. The DTP is the position immediately after the OAC. Up to four OACs can be specified.

Count Detection (CNT)

With the CNT option, the system will send a pre-defined number of digits (up to fifteen) before digit outpulsing is halted and the DTD is reconnected. Digit counting is done either one digit at a time, or as a string if fast Tone and Digit Switch (TDS) outpulsing is set up.

Digit Sequence (DGTS)

With the DGTS option, a table of up to 245 entries could be created of unique one-to-four digit sequences where the DTD should be reconnected after.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under Dial Tone Detector (DTD) package 138.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Set data for Flexible busy/overflow time.

2 LD 56 – Create tone and ringing parameters for one or more customers.

LD 16 – Set data for Flexible busy/overflow time.

Prompt	Response	Description
...		
DTD	(NO) YES	Dial Tone Detection (is not) is to be performed on this route.
SCDT	(NO) YES	Secondary dial tone (will not) will be used on route.

LD 56 – Create tone and ringing parameters for one or more customers.

Prompt	Response	Description
...		
TYPE	FDTD	Flexible Dial Tone Detection data.

Feature operation

No specific operating procedures are required to use this feature.

Flexible Direct Inward Dialing

Prior to the introduction of the Flexible Direct Inward Dialing (FDID) feature, hotels were required to purchase a large number of DID numbers that matched the number of hotel guest rooms. These DID DNs must be coordinated with the local exchange and become permanent in the system.

The FDID feature allows hotels to assign a temporary DID number to a guest room using a Property Management System (PMS) or Background Terminal (BGD).

Please refer to *Hospitality Features: Description and Operation* (553-3001-353) for complete information.

Flexible Feature Codes

Contents

This section contains information on the following topics:

Feature description 351
Operating parameters 353
Feature interactions 354
Feature packaging 357
Feature implementation 358
Feature operation 364

Feature description

Flexible Feature Codes (FFCs) are user-defined numbers of up to four digits that can be used in place of existing Special Prefix (SPRE) codes. With DN Expansion (DNXP) package 150, Flexible Feature Codes (FFCs) can be up to seven digits long. The Flexible Feature Code (FFC) feature allows customers to define different dialing codes for different features. There is no limit to the number of FFCs per prompt as long as each one is unique.

This feature allows the use of digits 0 through 9, and the asterisk (*) and octothorpe (#) to activate features. Special Prefix (SPRE) dialing feature is still supported, with or without the FFC feature enabled. However, the Special Prefix (SPRE) must be assigned in LD 15 in order for FFCs to operate for those features that also use SPRE codes.

The FFC package allows analog (500/2500 type) telephones to activate these features:

- Automatic Wake Up (AWU)
- Electronic Lock (ELK)
- Override, and
- Remote Call Forward (RCFW).

Customers define one or more codes at their discretion in LD 57 (FFC). For Service Change updates, refer to *Software Input/Output: Administration* (553-3001-311).

The basic FFC operation allows a telephone to access features normally available by dialing SPRE codes. FFCs are not supported, however, on a Meridian 1 proprietary telephone that is attempting a call pickup on a Dial Intercom ringing call.

A telephone can access a feature using FFC only if that telephone can currently access the same feature using SPRE dialing.

Any telephone, that does not currently have SPRE access, receives intercept treatment when dialing FFCs. Telephone operation remains the same (only the codes are different) so that the FFC code is dialed instead of the SPRE code. Therefore, each feature enabled must have an FFC individually defined.

When FFCT is YES in LD 57, the system returns a confirmation tone to the user after completing some feature operations.

The confirmation tone is the same as the special dial tone.

FFC allows analog (500/2500 type) telephones to Override established calls, based on the telephone's programmed Class of Service. Analog (500/2500 type) telephones can also activate and deactivate Call Forward by dialing a single FFC.

The confirmation Tone for FFC allows analog (500/2500 type) telephones and Meridian 1 proprietary telephones to receive a special tone when certain functions are complete. Confirmation Tone is returned following these events:

- Automatic Wake Up (any function)
- Call Forward (deactivate)

- Electronic Lock (any function)
- Ring Again (activate or deactivate)
- Room Status (any function)
- Speed Call Controller (add to Speed Call list), and
- Store Number (erase).

Confirmation Tone for FFC is returned when a predefined string is used as the end-of-dialing indicator for the following activities:

- Call Forward (activate)
- Permanent Hold (any function)
- Speed Call (store)
- Store Number (store), and
- Flexible Feature Code (any verification).

Confirmation Tone is provided for Speed Call store after the End-of-Dial string, such as the octothorpe (#), is entered.

Operating parameters

The SPRE feature must exist in order for FFC to operate.

The FFCs selected must be unique numbers up to seven digits long. They cannot conflict with any Directory Number (DN) already in the dialing plan.

LD 57 can allow no more than 100 FFCs to be modified in a single pass through Service Change.

Customers using the octothorpe (#) as part of their dialing plan can use a predefined string of digits for end-of-dialing indicators.

Changes to the Station Control Passwords (SCPWs) do not take affect until after a data dump and SYSLOAD. Configuring the system or enabling the feature changes SCPL = 0 in LD 15 to any length. This change takes effect immediately. Any other change to SCPL in LD 15 requires a data dump and SYSLOAD before taking effect. When the Station Control Password Length (SCPL) is changed, all associated passwords change accordingly at the next data dump and SYSLOAD. Changing SCPL from three to five automatically inserts leading zeros before all existing three-character passwords. Conversely, changing SCPL from five to three automatically truncates the leading characters of all existing five-character passwords.

Feature interactions

Attendant Blocking of Directory Number

If a Flexible Feature Code is dialed after pressing the Semi-automatic Camp-on (SACP) key to initiate an Attendant Blocking of DN attempt, overflow tone will be provided and the attempt cancelled.

Automatic Wake Up

Telephones can activate Automatic Wake Up (AWU) features for their own station with Common Controlled Switching Arrangement Class of Service.

The Automatic Wake Up feature may be active at the same time as Multiple Wake Up.

The attendant query function is not supported for Multiple Wake Up.

Multiple Wake Up from Attendant Consoles is not supported.

The Background Terminal (BGT) is not supported for Multiple Wake Up.

If one Automatic Wake Up time has been set using the Automatic Wake Up Activate (AWUA) FFC, only three additional Multiple Wake Up calls may be entered using the Multiple Wake Up Activate (MWUA) FFC.

Call Forward All Calls

When FFC is configured for a customer, #1 automatically becomes the FFC DN for both Call Forward Activate (CFWA) and Call Forward Deactivate (CFWD). When the same DN is used for both CFWA and CFWD, FFC toggles the Call Forward activated/deactivated state of the telephone. When Call Forward is activated for a telephone, entering #1 automatically deactivates Call Forward, no matter what follows #1. When Call Forward is deactivated for a telephone, the result of entering #1 depends on what follows:

- If the telephone goes on hook immediately, Call Forward is activated for the telephone to its previous call forward number.
- If a valid DN is entered after #1, call forward is activated for the telephone to that valid DN.
- If an invalid DN is entered after #1, call forward remains deactivated for the telephone.

Call Forward
Attendant and Network-Wide Remote

If the Outpulsing of Asterisk and Octothorpe (OPAO) package is equipped, the octothorpe (#) is treated as a dialed digit and does not signal the end of dialing. From one to three end-of-entry characters are defined in LD 15.

Call Pickup
Call Pickup, Directed

FFC codes are not supported on a Meridian 1 proprietary telephone during an attempt to pick up a Dial Intercom ringing call.

China – Flexible Feature Codes - Outgoing Call Barring

Flexible Feature Codes containing a “*” or an “#” will always be allowed by Outgoing Call Barring (OCB). Therefore, FFCs which can be used to make a call should be entirely numeric if barring of them is required.

Some FFCs are equivalent to Special Prefix functions and these will be subject to barring based on the equivalent Special Prefix codes, even if the FFC is entirely numeric.

Controlled Class of Service

If Electronic Lock (ELK) is activated, the CCRS Class of Service is used whether Controlled Class of Service (CCOS) is active or not. ELK takes precedence over CCOS. If ELK is deactivated, the set is treated as per existing operation.

When FFC ELKA and a password is entered, this set will use the CCRS Class of Service configured in LD 15. The CCRS Class of Service will always be used whether or not CCOS is currently controlling the set's Class of Service. When FFC ELKD and a password is entered, the set will use the appropriate Class of Service associated with this set. If CCOS is enabled for the set, the associated customer Class of Service is used (that is, CCRS, ECC1, or ECC2). If CCOS is not enabled for this set, the set's own Class of Service is used.

When FFC ELK is deactivated, the set reverts back to the Class of Service as it should be without FFC ELK, instead of always reverting back to the set's Class of Service (that is, if CCOS is enabled, it will use the customer's Class of Service; if CCOS is not enabled, it will use the set's Class of Service).

Intercept Treatment

If Intercept Treatment has been specified for a call to a vacant number (CTVN), the Digit Display (DDs) on the Attendant Console is affected by Flexible Feature Codes (FFCs). If no FFC has been defined, the dialed digits are displayed up to and including the first digit that fails to match any Directory Number (DN). If one or more FFCs have been defined, the dialed digits are displayed, up to and including the first digit that fails to match any FFC.

ISDN QSIG/EuroISDN Call Completion

Analog (500/2500 type) set can use Flexible Feature Codes (FFCs) to activate Call Completion to Busy Subscriber requests.

Pretranslation

Flexible Feature Codes must be accessible through a Pretranslation Table entry in order for users to activate features in this manner.

The Flexible Feature Code (FFC) feature will not be affected if the FFCs begin with "*" or "#", since before translation begins if the first digit is an "*" or "#" pretranslation will not be done. If any digits follow the FFC code, the first of the digits that follows will be pretranslated.

Special Prefix

Users are still able to use Special Prefix (SPRE) dialing (if the feature is enabled) with or without FFC defined.

Speed Call, System

With Flexible Feature Code (FFC), a confirmation tone is provided for Speed Call store after the end-of-dial (EOD) string is entered.

Feature packaging

Flexible Feature Codes (FFC) package 139 requires Controlled Class of Service (CCOS) package 81 only if Electronic Lock (ELK) is desired.

In addition, the SPRE dialing feature must be enabled for FFC functions.

2500 Telephone Features (SS25) package 18, and 500 Set Dial Access to Features (SS5) package 73 are required to support the following features:

- Call Forward
- Speed Call Controller
- Speed Call User
- Permanent Hold
- Call Park, and
- System Speed Call.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Set parameters for Flexible Feature Code.

2 LD 10 – Set Station Control Password Length for analog (500/2500 type) telephones.

3 LD 11 – Set Station Control Password Length for Meridian 1 proprietary telephones.

4 LD 57 – Define numbers for Flexible Feature Code.

LD 15 – Set parameters for Flexible Feature Code.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CCS	Controlled Class of Service options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CCRS	aaa	Controlled Class of Service (CCOS) (assigned when Electronic Lock (ELK) is activated), where: aaa = UNR (Unrestricted), TLD (Toll Denied), CTD (Conditionally Toll Denied), CUN (Conditionally Unrestricted), SRE (Semi-restricted), FRE (Fully Restricted), FR1 (Fully Restricted Level 1), FR2 (Fully Restricted Level 2).
TYPE	FFC	Flexible Feature Code Options.
- SCPL	x	Station Control Password Length (SCPL), 0-8. Entering 0 disables ELK and Remote Call Forward (RCFW) features at next data dump and SYSLOAD.
- FFCS	(NO) YES	(Do not) change FFC end-of-dialing indicator.

- - STRL	x	String length 1-3 (prompted only if FFCS = YES).
- - STRG	aaa	Character string to be used (up to string length; prompted only if FFCS = YES).

LD 10 – Set Station Control Password Length for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCPW	xx...xx X	Station Control Password (must be same length as SCPL in LD 15; enter X to delete password).
CLS	CCSA	Enable CCOS for Electronic Lock (ELK) and Remote Call Forward (RCFW).

LD 11 – Set Station Control Password Length for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

SCPW	xx...xx X	Station Control Password. Must be the same length as SCPL in LD 15. Enter X to delete the password. Delete the password only if SCPL = 0; otherwise receive an error code for no password to fit the SCPL.
CLS	CCSA	Enable CCOS for ELK and Remote Call Forward (RCFW).

LD 57 – Define numbers for Flexible Feature Code.

Prompt	Response	Description
REQ	NEW CHG OUT	Build new FFC data block, change FFC data block, remove FFC code.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	FFC Confirmation Tone.
CEPT	(NO) YES	Conférence Européen des Postes Tel defaults are (not) allowed, to be defined (prompted only if REQ = NEW).
REP*	n <CR>	Single-character replacement for * and # in CEPT defaults. Create defaults only.
ALL	(NO) YES	(Do not) remove all FFCs (prompted only if REQ = OUT).
CODE	aaaa ALL <CR>	FFC type. All prompts. No prompts.
- ASRC	xxxx	Automatic Set Relocation code.
- ATDA	xxxx	Autodial Activate code.
- ATDD	xxxx	Autodial Deactivate code.
- AUTH	xxxx	Authorization Code.

- AWUA	xxxx	Automatic Wake Up Activate code.
- AWUD	xxxx	Automatic Wake Up Deactivate code.
- AWUV	xxxx	Automatic Wake Up Verify code.
- CDRC	xxxx	Call Detail Recording Charge Account code.
- CFHO	xxxx	Call Forward/Hunt Override code.
- CFWA	xxxx	Call Forward All Calls Activate code.
- CFWD	xxxx	Call Forward All Calls Deactivate code.
- CFWV	xxxx	Call Forward All Calls Verify code.
- COND	xxxx	Conference Diagnostics code.
- CPAC	xxxx	Park Access Call code.
- CPRK	xxxx	Park Call code.
- CSHF	xxxx	Centrex Switchhook Flash code.
- C6DS	xxxx	Six-Party Conference code.
- CWGA	xxxx	Call Waiting Activate code.
- CWGD	xxxx	Call Waiting Deactivate code.
- DEAF	xxxx	Deactivate Ring Again and FWD codes.
- DPVS	xxxx	Data Port Verification code.
- ELKA	xxxx	Electronic Lock Activate code.
- ELKD	xxxx	Electronic Lock Deactivate code.
- GRPF	xxxx	Group Call code.
- GRCL	xxxx	Group Call List number.
- HOLD	xxxx	Permanent Hold code.

- ICFA	xxxx	Internal Call Forward Activate code.
- ICFD	xxxx	Internal Call Forward Deactivate code.
- ICFV	xxxx	Internal Call Forward Verify code.
- IMS	xxxx	Integrated Message System Access code.
- LILO	xxxx	Log-in, Log-out code for analog (500/2500 type) ACD telephones.
- MNTC	xxxx	Maintenance Access code.
- MSBA	xxxx	Make Set Busy Activate code.
- MSBD	xxxx	Make Set Busy Deactivate code.
- MTRC	xxxx	Malicious Call Trace code.
- MWRA	xxxx	Multiple Wake Up Repeat Activate code.
- MWUA	xxxx	Multiple Wake Up Activate code.
- MWUD	xxxx	Multiple Wake Up Deactivate code.
- NRDY	xxxx	Not Ready Activate or Deactivate code for analog (500/ 2500 type) ACD telephones.
- OVRD	xxxx	Override/Priority Override code.
- PUDN	xxxx	Pick Up Directory Number code.
- PUGR	xxxx	Pick Up Group code.
- PURN	xxxx	Pick Up Ringing Number code.
- RCFA	xxxx	Remote Call Forward Activate code.
- RCFD	xxxx	Remote Call Forward Deactivate code.
- RCFV	xxxx	Remote Call Forward Verify code.
- RDLN	xxxx	Redial Last Number code.

- RDNE	xxxx	Redial Number Erase code.
- RDSN	xxxx	Redial Saved Number code.
- RDST	xxxx	Redial Store code.
- RGAA	xxxx	Ring Again Activate code.
- RGAD	xxxx	Ring Again Deactivate code.
- RGAV	xxxx	Ring Again Verify code.
- RMST	xxxx	Room Status code.
- SADS	xxxx	Scheduled Access Restriction Disable code.
- SAEN	xxxx	Scheduled Access Restriction Enable code.
- SALK	xxxx	Scheduled Access Restriction Lock code.
- SAUN	xxxx	Scheduled Access Restriction Unlock code.
- SCPC	xxxx	Station Control Password Change code.
- SPCC	xxxx	Speed Call Controller code.
- SPCU	xxxx	Speed Call User code.
- SSPU	xxxx	System Speed Call User code.
- TFAS	xxxx	Trunk Answer from Any Station code.
- TRMD	xxxx	Terminal Diagnostics code.
- TRVS	xxxx	Trunk Verification code.
- USCR	xxxx	User Selectable Call Redirection.
- USTA	xxxx	User Status code.

Feature operation

For some features, the user can dial a different FFC to activate or deactivate a feature or to verify some feature operations. The tone for each event (activate, deactivate, verify) is the same as the default Confirmation Tone (special dial tone).

The Electronic Lock and Remote Call Forward FFCs are described here because Electronic Lock is packaged with Flexible Feature Codes and affects Remote Call Forward.

For information about using FFCs for other features, see the individual feature descriptions.

Electronic Lock

Electronic Lock (ELK), packaged with FFC, provides an SCPW for changing the status from the telephone. The SCPW also protects against changes to the Remote Call Forward (RCFW) feature. Entering a password length of 0 in LD 15 (SCPL) disables password control for both ELK and RCFW. Operating ELK requires enabling CCOS package 81.

To change the Class of Service from a telephone:

1 Dial the Electronic Lock Activate (ELKA) code.

2 Dial the SCPW. The telephone's Class of Service is changed to the CCRS value defined in LD 15.

To return the telephone to the originally defined Class of Service:

1 Dial the Electronic Lock Deactivate (ELKD) code.

2 Dial the SCPW. The telephone's Class of Service is changed to the values defined in LD 10 and LD 11.

Because the Class of Service defined for CCRS in LD 15 is usually lower than the Class of Service defined in LD 10 or LD 11, the Class of Service for a telephone is lowered by dialing the Electronic Lock Activate (ELKA) FFC and the password associated with that telephone. The user can activate from a remote telephone by dialing the ELKA FFC, the SCPW and the Directory Number to be changed. The same operation can deactivate the feature, using the Electronic Lock Deactivate (ELKD) code programmed in LD 57.

ELK operation has the following requirements:

- CCOS allowed, with CCSA Class of Service in LD 10 and LD 11, and CCRS defined in LD 15
- Set the password length in LD 15, at the SCPL prompt
- Add passwords in LD 10 and LD 11, at the SCPW prompt, and
- FFCT = YES in LD 57.

To change the SCPW for ELK:

1 Select a free extension.

2 Dial the SCPC code.

3 Dial the SCPW for your telephone.

4 Dial the new password.

5 To confirm, dial the new password again.

6 Hang up or press **Rls**.

Flexible Feature Code Boss Secretarial Filtering

Contents

This section contains information on the following topics:

Feature description 367
Operating parameters 367
Feature interactions 368
Feature packaging 370
Feature implementation 370
Feature operation 371

Feature description

The Flexible Feature Code Boss Secretarial Filtering (FFCSF) feature allows a set, designated as a “secretary” set, to filter calls coming in to a “boss” set. A boss or secretary set can be any Meridian 1 proprietary set or 16-button Dual-tone Multifrequency (DTMF) set. Filtering is a form of call screening, in which the calls coming into the boss set are presented to the secretary set to be answered and possibly transferred back to the boss set.

A boss set can have only one secretary set, while a secretary set can have an unlimited number of boss sets.

Operating parameters

A set cannot simultaneously be configured as a boss set and a secretary set.

The FFCSF Flexible Feature Code must be unique and not conflict with the customer dialing plan.

Secretary DNs which are programmed on a boss set cannot already be part of the customer's DN plan, nor conflict with it.

The FFCSF feature cannot be applied to sets having Multiple Appearance DNs.

In a networking environment, a boss set and secretary set must be on the same node.

Easy Change (ECHG) requests cannot be made against the Secretarial Filtering (SFLT) and Secretarial Forwarding DN (SFDN) prompts in LDs 10 and 11.

Feature interactions

Attendant Blocking of Directory Number

The FFC Boss Secretary Filtering feature will be overridden. If an Attendant Blocking of DN attempt is made for a set that has the Boss Secretary Filtering feature active, the dialed DN will be blocked if idle. If it is busy, busy tone will be heard.

Attendant-Extended Calls

Attendant-extended third-party calls to a boss set will be subject to filtering if filtering on the boss set is active for all calls. If filtering is allowed for external calls only, the attendant will be filtered only if the third party is external.

Call Forward All Calls

Although Call Forward All Calls and FFCSF can be equipped on the same set, they cannot both be active at the same time. There is no precedence of one over the other; it is not possible to activate one if the other is active on the set.

Call Forward Busy
Call Waiting

A Call Forward Busy or Call Waiting to a boss set with filtering active is routed to the secretary set.

Call Forward and Busy Status

If the secretary set is a Meridian 1 proprietary telephone, or a compact digital set, it can be equipped with a Call Forward and Busy Status (BFS) key/lamp pair, to perform the following:

- monitor the status of the Call Forward feature on a boss set
- activate/deactivate the Call Forward feature on a boss set
- monitor whether or not a boss set is busy on a call, and
- override the Call Forward All Calls feature on a boss set, in order to place a call to the boss set.

The above functions, however, can only be performed by the secretary set while it is in an unattended state, since BFS and FFCSF cannot be active simultaneously.

Camp-On

When an attendant is attempting to Camp-on a call to a boss set with filtering active, the call is routed to the secretary set, if the filtering is active for all calls. If filtering is active for external calls only, the call is routed to the secretary set if the call is an external call.

Hot Line
Private Line

FFCSF takes precedence over Private Line and Hot Line.

Hunting

A boss set with filtering activated is passed over by Hunting; the next hunt sequence is to the secretary set.

Lockout, DID Second Degree Busy, and MFE Signaling Treatment

Flexible Feature Code Boss Secretarial Filtering takes precedence over lockout and second degree busy.

Network Intercom

Hot Type I calls override this feature (for instance, Hot Type I calls are not filtered by FFC Boss Secretarial filtering). The call terminates on the Boss' set and is not forwarded to the secretary.

FFC Boss Secretarial Filtering takes precedence over enhanced Hot Type D calls. In this case, if FFC Boss Secretarial Filtering is active, calls terminate on the secretary's set.

Voice Call

A call to a Voice Call key on a boss set with filtering active is not filtered to the secretary set.

Feature packaging

This feature is packaged under Boss Secretarial Filtering (FFCSF), package 198.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 or LD 11 – Respond to the Secretarial Filtering (SFLT) prompt.

2 LD 57 – Define the Secretarial Filtering Access Code.

LD 10 or LD 11 – Respond to the Secretarial Filtering (SFLT) prompt.

Prompt	Response	Description
...		
SFLT	(NO) BOSS SEC	Secretarial Filtering, prompted with Boss Secretarial Filtering (FFCSF) package 198. Designate a telephone set entering either BOSS for boss set, SEC for secretary set, or NO for no designation. SEC, (NO), and <CR> take you to the next prompt.

- SFDN	xxxx	Secretarial Forwarding DN of secretary set to which filtered calls should be forwarded, prompted if response to SFLT = BOSS.

LD 57 – Define the Secretarial Filtering Access Code.

Prompt	Response	Description
...		
SFAC	xxxx	Secretarial Filtering Access code.

Feature operation

The FFCSF feature may be accessed from the boss set and secretary set using the same Flexible Feature Code (FFC) followed by a control digit.

On a boss set, the following control digits can be dialed:

- 7, to activate filtering for all external calls
- 8, to activate filtering for all external and internal calls, and
- 9, to cancel filtering.

Confirmation tone is given to the boss set after filtering has been successfully activated or deactivated, or if filtering was already activated. Afterwards, a special dial tone (the same as the one used to indicate that Call Forward is active on a set) is provided to the boss set whenever it goes off-hook, as an audible reminder that the feature is active.

If filtering could not be activated by the boss set due to one of the following conditions, overflow tone is returned:

- the secretary set assigned to the boss set is not attended, or
- Call Forward All Calls is active on the boss set.

On a secretary set, the following control digits may be dialed:

- 8 – to place the secretary set in attended state, allowing calls to be filtered to it from a boss set.
- 9 – to place the secretary set in unattended state and to disable the boss set filtering.

In either case, confirmation tone is returned to the secretary set.

Flexible Key Assignment

Contents

This section contains information on the following topics:

Feature description 373
Operating parameters 373
Feature interactions 373
Feature packaging 374
Feature implementation 374
Feature operation 374

Feature description

The Flexible Key Assignment feature allows the assignment of features other than volume up, volume down, and hold to the three keys located below the dial pad on an SL-1 telephone. Any feature not requiring a lamp indicator can be assigned to these keys.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packed under International Supplementary Features (SUPP), package 131.

Feature implementation

LD 11 – Assign key functions to keys.

Prompt	Response	Description
...		
KEY	xx aaa	Key number (0-9) and key function.

Feature operation

Press the appropriate key on the SL-1 telephone to activate the feature assigned to it.

Flexible Orbiting Prevention Timer

Contents

This section contains information on the following topics:

Feature description 375
Operating parameters 375
Feature interactions 376
Feature packaging 376
Feature implementation 376
Feature operation 376

Feature description

The Orbit Prevention feature prevents an infinite loop from being created in a network-wide Call Forward configuration resulting from set A being call forwarded (all calls) to set B at another node, which in turn has been call forwarded back to set A. A check is provided through the Flexible Orbiting Prevention Timer (FOPT) that prohibits any set from call forwarding more than one call off-node for a period of 14 seconds.

The Orbit Prevention feature allows the Flexible Orbiting Prevention Timer (FOPT), to be service changeable from 0 to 30 seconds (even numbers only). If a value of 0 is defined, then Orbit Prevention is disabled and call forwarding is not inhibited in any way.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Enter an even value between 0-30 seconds, at the FOPT prompt to define the Orbit Prevention Timer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	NET	ISDN and ESN Networking Options.
...		
- FOPT	0-(14)-30	Flexible Orbiting Prevention Timer. The number of seconds in two-second intervals that CFW should be suspended on a set that has just forwarded a call off node. If an odd number is entered, the number is rounded up to the next even number, and the message “FOPT ROUNDED TO xx” is printed.

Feature operation

No specific operating procedures are required to use this feature.

Flexible Tone and Digit Switch Control

Contents

This section contains information on the following topics:

Feature description 377
Operating parameters 379
Feature interactions 379
Feature packaging 379
Feature implementation 380
Feature operation 381

Feature description

This feature allows the system to generate the many tones and cadences required for call processing in various countries. The system must be equipped with Flexible Tone and Digit Switch (TDS) circuit packs. One TDS pack is inserted in each network shelf in place of a network circuit pack.

The TDS packs are pre-overlay programmed with certain basic tone characteristics (frequencies, levels and cadences) which are then combined in various ways to produce the following tones:

- ACD ring-again ringback tone
- busy tone
- call forward dial tone
- call forward message-waiting dial tone

- camp-on confirm tone
- control dial tone
- dial tone
- dial-0 recall tone
- hold confirmation tone
- listed DN tone
- message waiting dial tone
- overflow tone
- preemption tone
- ringback tone
- test tone

These tones are also service changeable in LD 56. When call processing requires a particular tone, software sends the code defining that tone to the TDS pack. The TDS pack then generates the tone.

A number of other tones and associated cadences are available from the TDS but are assigned by software in LD 56. These are:

- agent observe tone
- call waiting tone
- intrusion tone
- override tone

The following tones are likely to be defined as bursts, but are still software controlled:

- ATV completion busy tone
- observe blocking tone
- off-hook queuing tone
- set relocate tone
- telset messaging alert tone

- telset messaging OK tone
- telset status update tone

Three exceptions to the categories of tones described so far are special dial tone, expensive route warning tone, and precedence call waiting tone. These tones are flexible only in their sound and not in their cadence.

Also included are distinctive or precedence ringing for analog (500/2500-type), M1000-series, SL-1, and digital telephones. Refer to LD 56 in *Software Input/Output: Administration* (553-3001-311) for the identification of these tones and cadences.

The tone and ringing requirements of the customer determine which TDS is required.

This feature also provides the following:

- an additional make/break ratio is available for ten pulses per second dialpulsing
- variable inter-digit pause time is flexible and can be assigned in LD 56 for digitone and dialpulse digits
- two additional DTMF outpulsing rates are available and assigned in LD 17

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under Flexible Tones and Cadences (FTC) package 125.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Modify the system hardware and software parameters.

2 LD 56 – Modify or change customer's tone and ringing parameters.

LD 17 – Modify the system hardware and software parameters.

Prompt	Response	Description
...		
PARM	(NO) YES	Change system parameters.
ABCD	(NO) YES	16-Button DTMF operation is (is not) enabled.
	<CR>	Original value is left unchanged.
DTRB	100	100 millisecond bursts of DTMF with 100 millisecond interdigit pause.
	50	50 millisecond bursts of DTMF with 50 millisecond interdigit pause.
	60	60 millisecond bursts of DTMF with 90 millisecond interdigit pause.
	70	70 millisecond bursts of DTMF with 70 millisecond interdigit pause.

LD 56 – Modify or change customer's tone and ringing parameters.

Prompt	Response	Description
...		
TYPE	FTC	Flexible tone and ringing.

Feature operation

No specific operating procedures are required to use this feature.

Flexible Trunk to Trunk Connections

Contents

This section contains information on the following topics:

Feature description 383
Operating parameters 398
Feature interactions 400
Feature packaging 403
Feature implementation 403
Feature operation 405

Feature description

The Flexible Trunk to Trunk Connections (FTT) feature controls trunk to trunk connections for Transfer, Supervised Conference, and unsupervised Conference, based upon the Station's Class of Service. This feature is used with or without the Trunk Barring (TBAR) feature. The Flexible Trunk to Trunk Connections feature provides the following options at a set level:

- allows trunk to trunk connections for Transfer and Conference
- denies trunk to trunk connections for Transfer and Conference
- allows trunk to trunk connections for Supervised Conference only, and denies trunk to trunk connections for Transfer and unsupervised Conference

The Conference feature allows additional parties to join an established call. One internal Directory Number must always be involved in the Conference call for a Supervised Conference. A system user can conference two or more trunks and then drop out of the conference, leaving the other trunks connected. This is an unsupervised Conference.

When Flexible Trunk to Trunk Connections is used in conjunction with the Trunk Barring feature, **one** of the following options may be selected:

- Additional set level restrictions can be added to the existing Customer level Trunk Barring.
- The restrictions placed by Trunk Barring, based upon the set's Flexible Trunk to Trunk Connections Class of Service, can be lifted.
- All set based trunk to trunk connections can be controlled for Conference and Transfer, depending upon the set's Flexible Trunk to Trunk Connections Class of Service, whether or not the route is barred by TBAR.

The functionality of the Flexible Trunk to Trunk Connections feature is activated by Flexible Trunk to Trunk Connections Options (FTOP prompt) in the Customer Data Block and controlled by the Station's Class of Service. The options that are available at a Customer level are dependent upon whether or not Trunk Barring (package 132) is configured.

Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured

When Flexible Trunk to Trunk Connections is used without Trunk Barring configured, the following Classes of Service are provided at a set level:

- When CLS = FTTU, Flexible Trunk to Trunk Connections Unrestricted, trunk to trunk connections are allowed for both Conference and Transfer. Flexible Trunk to Trunk Connections Unrestricted (FTTU) is the default value.

- When CLS = FTTR, Flexible Trunk to Trunk Connections Restricted (FTTR), trunk to trunk connections are denied for both Conference and Transfer.
- When CLS = FTTC, Flexible Trunk to Trunk Connections Conditional, trunk to trunk connections are allowed for Supervised Conference. Trunk to trunk connections are denied for Transfer and unsupervised Conference. Flexible Trunk to Trunk Connections Conditional (FTTC) is the default for new sets.

When Flexible Trunk to Trunk Connections is used without Trunk Barring configured, the following Flexible Trunk to Trunk Connections Options are available at a Customer level:

- When FTOP = FRES, Flexible Trunk to Trunk Connections Restricted, the Flexible Trunk to Trunk Connections feature does not function. The customer can still configure the set's Class of Service; however, the Class of Service does not take effect. Flexible Trunk to Trunk Connections Restricted (FRES) is the default value.
- When FTOP = FTLY, Flexible Trunk to Trunk Connections Only, trunk to trunk connections are controlled exclusively by the Flexible Trunk to Trunk Connections feature, based upon the set's Class of Service.

Figure 6 illustrates the functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured.

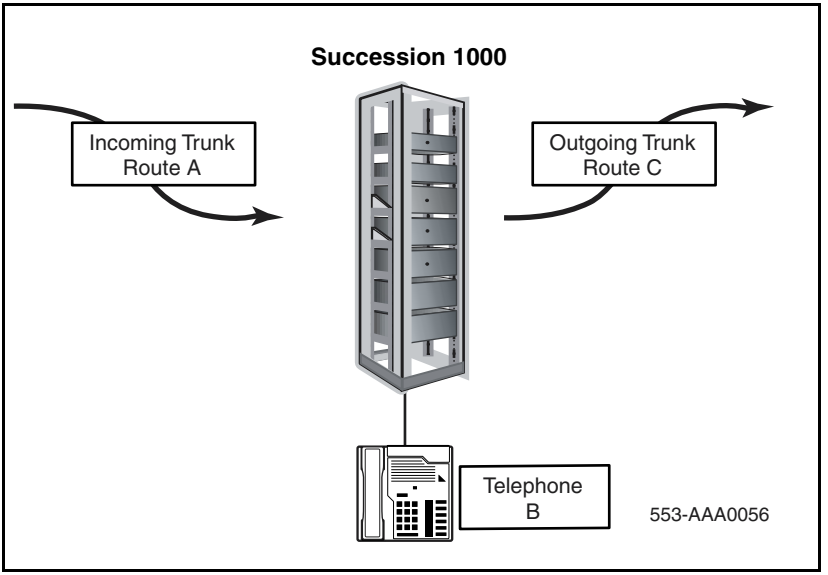
In Figure 6, Set B is established with Trunk Route A and initiates a transfer or a conference with Trunk Route C.

Flexible Trunk to Trunk Connections Options (FTOP) = Flexible Trunk to Trunk Connections Only (FTLY)

Referring to Figure 6, when Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (**FTLY**) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Unrestricted (**FTTU**), the following is true:

- Telephone B can complete the Call Transfer between Trunk Routes A and C, as long as no other restrictions apply.
- Telephone B can conference Trunk Routes A and C and then disconnect. In this case, Trunk Routes A and C remain connected, as long as no other restrictions apply.

Figure 6
Functionality of Flexible Trunk to Trunk Connections without Trunk Barring configured



When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (**FTLY**) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Restricted (**FTTR**), the following is true:

- Telephone B cannot transfer Incoming Trunk Route A to Outgoing Trunk Route C.
- Telephone B cannot complete the conference involving Trunk Routes A and C.

With Class of Service set to Flexible Trunk to Trunk Connections Restricted (FTTR), a consultation connection initiated by telephone B to Trunk Route C is not affected by Flexible Trunk to Trunk Connections.

Referring to Figure 6, when Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (**FTLY**) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Conditional (**FTTC**), the following is true:

- Telephone B cannot complete the Call Transfer from Trunk Route A to Trunk Route C.
- Telephone B can complete the Supervised Conference with Trunk Routes A and C, as long as no other restrictions apply. If Set B drops out of this conference, Trunk Routes A and C are disconnected.

Table 11 is a matrix that summarizes the possible selections for Station Class of Service and Flexible Trunk to Trunk Connections Options available for Flexible Trunk to Trunk Connections without Trunk Barring configured.

Table 11
CLS and FTOP Matrix for Flexible Trunk to Trunk Connections without Trunk Barring configured

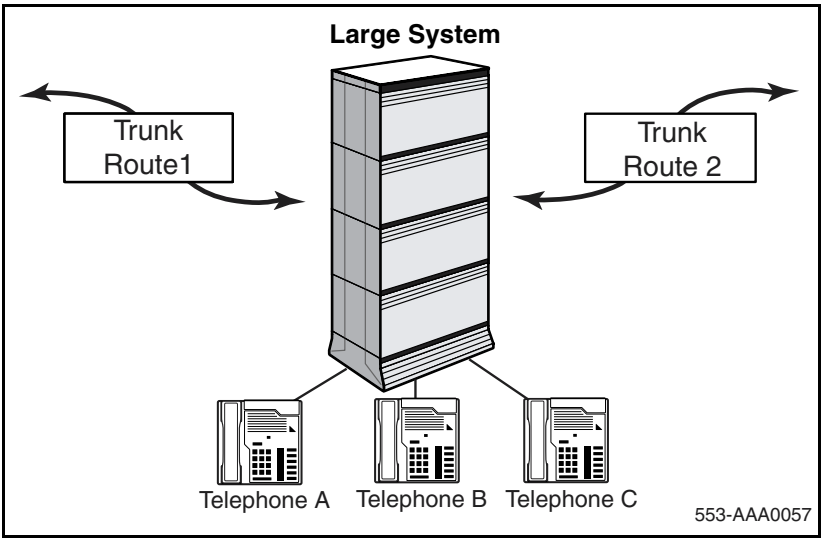
Station Classes of Service (LDs 10 and 11)	Customer Level Options (LD 15)	
	FTOP = FRES (Default)	**FTOP = FTLY**
CLS = FTTU (Default for existing sets)	No effect on Class of Service. Existing restrictions still apply.	Allows trunk to trunk connections for both Transfer and Conference.
CLS = FTTR	No effect on Class of Service. Existing restrictions still apply.	Blocks all trunk to trunk connections for Transfer and Conference.
CLS = FTTC (Default for new sets)	No effect on Class of Service. Existing restrictions still apply.	Allows trunk to trunk connections for Supervised Conference only. Denies trunk to trunk connections for Transfer and unsupervised Conference.

Functionality of Flexible Trunk to Trunk Connections for Supervised Conference

For Supervised Conference, at least one internal set must be involved in the conference. With the Flexible Trunk to Trunk Connections feature configured, if the last set that drops out of the conference has Class of Service set to Flexible Trunk to Trunk Connections Restricted (FTTR) or Flexible Trunk to Trunk Connections Conditional (FTTC), the call is disconnected. If the last set that drops out of the conference has Class of Service (CLS) set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), the call is not disconnected.

Figure 7 illustrates the functionality of Flexible Trunk to Trunk Connections for Supervised Conference.

Figure 7
Flexible Trunk to Trunk Connections for Supervised Conference



Referring to Figure 7, Telephones A, B, and C have Class of Service set to Flexible Trunk to Trunk Connections Unrestricted (FTTU), Flexible Trunk to Trunk Connections Restricted (FTTR), and Flexible Trunk to Trunk Connections Conditional (FTTC) respectively. Telephones A, B, and C and Trunk Routes 1 and 2 are involved in a conference.

- If A is the last internal telephone to drop out of the conference, the call is not disconnected by the Flexible Trunk to Trunk Connections feature, as Class of Service is set to Flexible Trunk to Trunk Connections Unrestricted (FTTU) for A. Other restrictions, however, may cause the call to disconnect. This is an unsupervised conference. The present functionality is maintained.
- If B is the last internal telephone to drop out of the conference, the call is disconnected, as Class of Service is set to Flexible Trunk to Trunk Connections Restricted (FTTR) for B.
- If C is the last internal telephone to drop out of the conference, the call is disconnected, as Class of Service is set to Flexible Trunk to Trunk Connections Conditional (FTTC) for C.

Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured

Trunk Barring provides the option of denying a direct or modified connection between Customer defined routes. Trunk Barring works with Route Access Restriction Tables (ARTs), as defined in LD 56.

When the Flexible Trunk to Trunk Connections feature is used with Trunk Barring (TBAR) configured, additional flexibility in controlling the trunk to trunk connections for Transfer and Conference is provided.

If Flexible Trunk to Trunk Connections is implemented with the Trunk Barring feature, the following four options are available at a customer level:

- When FTOP = FRES, Flexible Trunk to Trunk Connections Restricted, the Flexible Trunk to Trunk Connections feature does not function. The customer can still configure the set's Class of Service; however, the Class of Service does not take effect. Flexible Trunk to Trunk Connections Restricted (FRES) is the default value.

- When FTOP = TBFT, Trunk Barring Flexible Trunk to Trunk Connections, additional restrictions are applied, depending upon the set's Class of Service. Trunk to trunk connections barred by TBAR always remain restricted. Connections not barred by TBAR utilize the set's Class of Service.
- When FTOP = FTTB, Flexible Trunk to Trunk Connections Trunk Barring, Flexible Trunk to Trunk Connections lifts TBAR restrictions for routes barred by TBAR, based upon the set's Class of Service. Flexible Trunk to Trunk Connections does not apply any new restrictions for non-barred routes.
- When FTOP = FTLY, Flexible Trunk to Trunk Connections Only, trunk to trunk connections for Transfer or Conference that are on barred and non-barred routes are controlled exclusively by the Flexible Trunk to Trunk Connections feature.

The Flexible Trunk to Trunk Connections feature provides the same Class of Service options at a set level with or without Trunk Barring configured. (CLS = FTTU, FTTR, FTTC).

Flexible Trunk to Trunk Connections Options (FTOP) = Trunk Barring Flexible Trunk to Trunk Connections (TBFT)

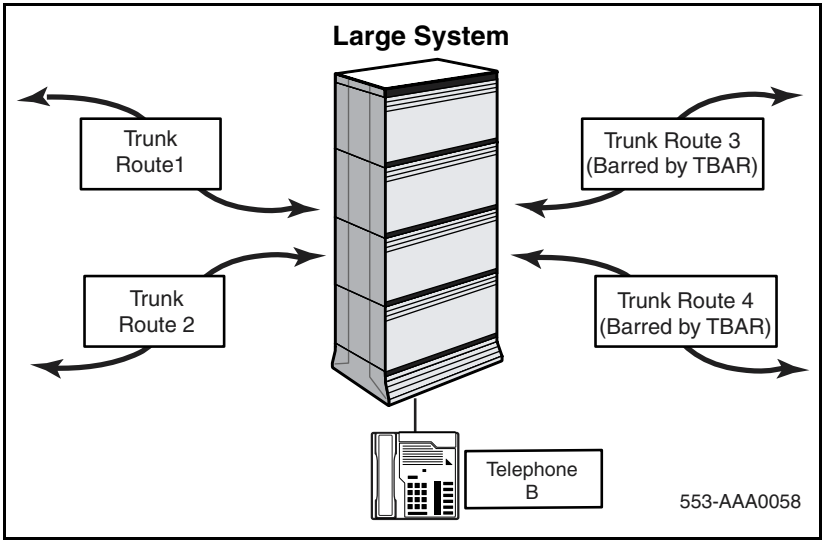
Figure 8 illustrates Flexible Trunk to Trunk Connections functionality with Trunk Barring configured and FTOP set to TBFT in LD 15.

In Figure 8, B is established on a call with Trunk Route 1. Trunk Routes 1 and 2 are not barred by TBAR, but Trunk Routes 3 and 4 are barred connection to any other route. Trunk Routes 1, 2, 3, and 4 are both incoming and outgoing.

Referring to Figure 8, when Flexible Trunk to Trunk Connections Options is set to Trunk Barring Flexible Trunk to Trunk Connections (**TBFT**) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Unrestricted (**FTTU**), the following is true:

- B can complete the Call Transfer between Trunk Routes 1 and 2.
- B can conference Trunk Routes 1 and 2 and then disconnect. In this case, Trunk Routes 1 and 2 remain connected, as TBAR does not bar the connection between the two trunks.
- B cannot complete Transfer or Conference from Trunk Routes 1 or 2 to Trunk Routes 3 or 4, as these trunk routes are barred by TBAR.

Figure 8
Functionality of Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = TBFT



Referring to Figure 8, when Flexible Trunk to Trunk Connections Options is set to Trunk Barring Flexible Trunk to Trunk Connections (**TBFT**) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Restricted (**FTTR**), the following is true:

- B cannot complete Transfer or Conference with Trunk Routes 1 and 2, even though the connectivity between the trunks is allowed by TBAR. This Class of Service functions as though the two trunks are blocked by the Trunk Barring feature.
- B cannot complete Transfer or Conference from Trunk Routes 1 and 2 to Trunk Routes 3 or 4, as these trunks are barred by TBAR.

Referring to Figure 8, when Flexible Trunk to Trunk Connections Options is set to Trunk Barring Flexible Trunk to Trunk Connections (**TBFT**) and the Class of Service of Set B is set to Flexible Trunk to Trunk Connections Conditional (**FTTC**), the following is true:

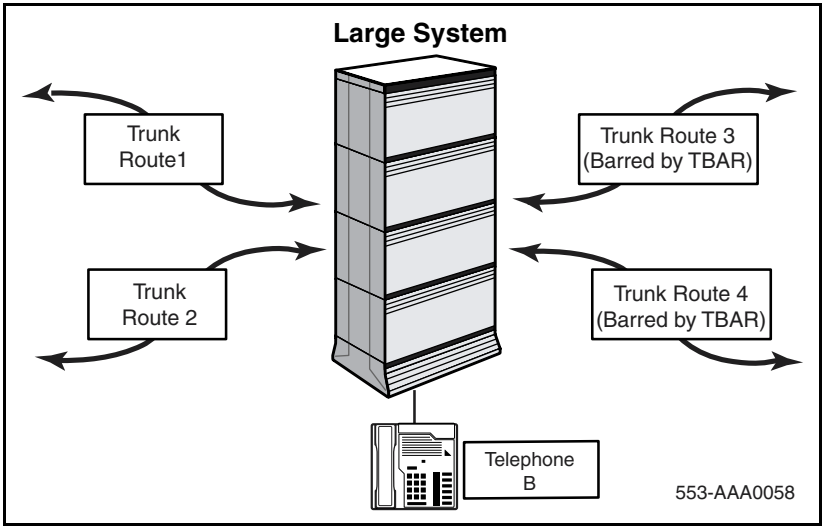
- B cannot complete the Call Transfer from Trunk Route 1 to Trunk Route 2

- B can complete the Supervised Conference with Trunk Routes 1 and 2. If Set B drops out of this conference, Trunk Routes 1 and 2 are disconnected.
- B cannot complete both Transfer and Conference from Trunk Routes 1 or 2 to Trunk Routes 3 or 4, as these trunks are barred by TBAR.

Flexible Trunk to Trunk Connections Options (FTOP) = Flexible Trunk to Trunk Connections Trunk Barring (FTTB)

Figure 9 illustrates Flexible Trunk to Trunk Connections functionality with Trunk Barring configured and FTOP set to FTTB.

Figure 9
Flexible Trunk to Trunk Connections with Trunk Barring configured and FTOP = FTTB



In Figure 9, Routes 1, 2, 3, and 4 are both incoming and outgoing. Access to different trunks is given as follows:

- From Trunk Route 1, connection is allowed to Trunk Routes 2, 3, and 4.
- From Trunk Route 2, connection is allowed to Trunk Routes 1, 3, and 4.

- From Trunk Route 3, connection is denied to Trunk Routes 1, 2, and 4.
- From Trunk Route 4, connection is denied to Trunk Routes 1, 2, and 3.

In short, any call from/to Trunk Route 1 or Trunk Route 2 is allowed. Any call from Trunk Route 3 and Trunk Route 4 is denied to all other trunk routes.

Referring to Figure 9, when the Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Trunk Barring (**FTTB**) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Unrestricted (**FTTU**), all TBAR restrictions for Transfer and Conference are lifted.

On a TBAR unrestricted trunk, B receives a call on incoming Trunk Route 1. The call is established. B initiates a call with any of the Trunk Routes 2, 3, or 4.

When TBAR does not restrict connection from Trunk Route 1 to any other trunk route:

- B can transfer the call on Trunk Route 1 to any of the Trunk Routes 2, 3, or 4.
- B can conference the call on Trunk Route 1 with any of the Trunk Routes 2, 3, or 4.

On a TBAR restricted trunk, B receives a call on incoming Trunk Route 3. The call is established. B initiates a call with any of the Trunk Routes 1, 2, or 4.

In this case, TBAR restricts connection from Trunk Route 3 to any other trunk route. However, as B has CLS set to FTTU, the TBAR restriction is lifted for B. Therefore:

- B can transfer the call on Trunk Route 3 to any of the Trunk Routes 1, 2, or 4
- B can conference the call on Trunk Route 3 with any of the Trunk Routes 1, 2, or 4.

Referring to Figure 9, when the Flexible Trunk to Trunk Connections Options (FTOP) is set to Flexible Trunk to Trunk Connections Trunk Barring (**FTTB**) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Restricted (**FTTR**), the existing TBAR functionality is retained.

On a TBAR unrestricted trunk, B receives a call on incoming Trunk Route 1, which is a TBAR unrestricted trunk. The call is established. B initiates a call with any of the Trunk Routes 2, 3, or 4.

When TBAR does not restrict connection from Trunk Route 1 to any other trunk route:

- B can transfer the call on Trunk Route 1 to any of the Trunk Routes 2, 3, or 4.
- B can conference the call on Trunk Route 1 with any of the Trunk Routes 2, 3, or 4.

On a TBAR restricted trunk, B receives a call on incoming Trunk Route 3. The call is established. B initiates a call with any of the Trunk Routes 1, 2, or 4.

In this case, TBAR restricts connection from Trunk Route 3 to any other trunk route. As B has Class of Service set to FTTR, the TBAR restriction is not lifted for this set.

- B cannot transfer the call on Trunk Route 3 to any of the Trunk Routes 1, 2, or 4.
- B cannot conference the call on Trunk Route 3 with any of the Trunk Routes 1, 2, or 4.

Referring to Figure 9, when the Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Trunk Barring (**FTTB**) and the Class of Service of B is set to Flexible Trunk to Trunk Connections Conditional (**FTTC**), TBAR restrictions for Supervised Conference are lifted. TBAR restrictions for Transfer and unsupervised Conference are maintained.

On a TBAR unrestricted trunk, B receives a call on incoming Trunk Route 1. The call is established. B initiates a call with any of the Trunk Routes 2, 3, or 4.

When TBAR does not restrict connection from Trunk Route 1 to any other trunk route:

- B can transfer the call on Trunk Route 1 to any of the Trunk Routes 2, 3, or, 4.
- B can conference the call on Trunk Route 1 with any of the Trunk Routes 2, 3, or 4.

On a TBAR restricted trunk, B receives a call on incoming Trunk Route 3. The call is established. B initiates a call with any of the Trunk Routes 1, 2, or 4.

When TBAR restricts connection from Trunk Route 3 to any other trunk routes:

- B cannot transfer the call on Trunk Route 3 to any of the Trunk Routes 1, 2, or 4.

However, as B has Class of Service set to FTTC, the TBAR restriction is lifted for Supervised Conference. Therefore:

- B can conference the call on Trunk Route 3 with any of the Trunk Routes 1, 2, or 4.
- Once B drops out of the conference, the two remaining TBAR trunks are disconnected.

Since all telephones that are already present in the system default to a Class of Service of Flexible Trunk to Trunk Connections Unrestricted (FTTU), when the Customer Option is changed to Flexible Trunk to Trunk Connections Trunk Barring (FTTB), TBAR restrictions for all telephones are lifted for Conference and Transfer. Therefore, the Class of Service must be changed to Flexible Trunk to Trunk Connections Restricted (FTTR), in order to maintain the existing TBAR functionality. The telephone sets that are new to the system default to a Class of Service of FTTC.

Flexible Trunk to Trunk Connections Options (FTOP) = Flexible Trunk to Trunk Connections Only (FTLY)

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (**FTLY**) and the Class of Service is set to Flexible Trunk to Trunk Connections Unrestricted (**FTTU**), trunk to trunk connections are allowed for both Conference and Transfer, irrespective of whether or not TBAR is activated.

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (**FTLY**) and the Class of Service is set to Flexible Trunk to Trunk Connections Restricted (**FTTR)**, trunk to trunk connections are denied for both Conference and Transfer, irrespective of whether or not TBAR is activated.

When Flexible Trunk to Trunk Connections Options is set to Flexible Trunk to Trunk Connections Only (**FTLY**) and the Class of Service is set to Flexible Trunk to Trunk Connections Conditional (**FTTC**), trunk to trunk connections are allowed for Supervised Conference only. Trunk to trunk connections for Transfer and unsupervised Conference are denied, irrespective of whether or not TBAR is activated.

Table 12 is a matrix that summarizes the possible selections for Station Class of Service and Flexible Trunk to Trunk Connections Options available for Flexible Trunk to Trunk Connections when Trunk Barring is configured.

Table 12
CLS and FTOP Matrix for Flexible Trunk to Trunk Connections with TBAR configured

Station Classes of Service (LDs 10 and 11)	Customer Level Options (LD 15)			
	FTOP = FRES (Default)	**FTOP = TBFT**	**FTOP = FTTB**	**FTOP = FTLY**
CLS = FTTU (Default for existing sets)	Existing TBAR functionality.	Existing TBAR functionality.	Lifts all TBAR restrictions for Transfer and Conference.	Allows Transfer and Conference, irrespective of whether or not TBAR is activated, unless other restrictions exist.
CLS = FTTR	Existing TBAR functionality.	Blocks all trunk to trunk connections for both Transfer and Conference.	Existing TBAR functionality.	Blocks all trunk to trunk connections for Transfer and Conference.
CLS = FTTC (Default for new sets)	Existing TBAR functionality.	Allows trunk to trunk connections for Supervised Conference on non-TBAR routes. Denies Transfer and unsupervised Conference on all trunk to trunk connections not blocked by TBAR.	Lifts TBAR restrictions for Supervised Conference only. Maintains TBAR restrictions for unsupervised Conference and Transfer.	Blocks trunk to trunk connections for Transfer and unsupervised Conference. Allows trunk to trunk connections for Supervised Conference.

Operating parameters

Flexible Trunk to Trunk Connections is configured at a set level, by defining the Class of Service (CLS) prompt in LDs 10 or 11.

All existing telephone sets default to a Class of Service of Flexible Trunk to Trunk Connections Unrestricted (FTTU) upon initial software conversion. When new telephone sets are added and configured, they default to a Class of Service of Flexible Trunk to Trunk Connections Conditional (FTTC).

In the Customer Data Block, Flexible Trunk to Trunk Connections Options can be set to Trunk Barring Flexible Trunk to Trunk Connections (TBFT) and Flexible Trunk to Trunk Connections Trunk Barring (FTTB) only when the Trunk Barring is configured. Flexible Trunk to Trunk Connections Options (FTOP) is set to the default, Flexible Trunk to Trunk Connections Restricted (FRES), to maintain the existing functionality.

A telephone set with a Class of Service of Flexible Trunk to Trunk Connections Restricted (FTTR) cannot initiate a Conference call to an outgoing trunk, although it can be included in a conference. If this type of telephone set is the last set to disconnect from the conference, the call is ended. The established trunks are released.

If a conference is on hold and an additional telephone set attempts to join the conference over a barred trunk route and through a telephone set that has Class of Service set to Flexible Trunk to Trunk Connections Restricted (FTTR), then Flexible Trunk to Trunk Connections does not permit a consultation connection. This is as per the existing operation.

If more than two trunks are involved in a call and all internal calls drop from the conference, Flexible Trunk to Trunk Connections does not affect the Conference disconnection.

Multiple Appearance, Single Call Arrangement DNs allow a single call to be active on the DN, regardless of its number of appearances. If the Single Call Ringing DN is established in a call, another appearance of the DN can enter into the call, if the Privacy feature is not in effect, by going off hook or by pressing the Multiple Appearance Single Call DN key. Flexible Trunk to Trunk Connections restrictions for Conference are applicable in such a case.

As per the existing operation, answer and disconnect supervision is a requirement for Transfer and Conference.

Flexible Trunk to Trunk Connections does not support Basic Rate Interface (BRI) telephone sets or Attendant Console operations.

Flexible Trunk to Trunk Connections supports Analog and ISDN trunks. R2MFC and AC15 signaling is also supported. Flexible Trunk to Trunk Connections does not support Service trunks, such as Recorded Announcement (RAN), Paging (PAG), Dictation (DIC), Music (MUS), and Automatic Wake Up Recorded Announcement (AWR).

Call Redirection features are not supported with Flexible Trunk to Trunk Connections.

With Flexible Trunk to Trunk Connections, unless the Trunk to Trunk Connection feature is configured, two outgoing trunk connections are blocked for Transfer and unsupervised Conference.

Customer Controlled Routing (CCR), Meridian Link, and Application Module Link (AML) applications are not affected by the Flexible Trunk to Trunk Connections feature.

> ***Note:*** When adding a new telephone set, the default Class of Service is Flexible Trunk to Trunk Connections Conditional (FTTC). This could impact an application's ability to conference or transfer a call to an outgoing trunk. In the case where this functionality is required, the Class of Service must be changed on the set to Flexible Trunk to Trunk Connections Unrestricted (FTTU).

Flexible Trunk to Trunk Connections blocks the initiation of Conference. Applications, such as Break In, Barge In, Bridging, and Overriding, are not supported.

Feature interactions

Access Restrictions

Access Restrictions limits terminal access to the exchange network, private network, and certain features and services. During the call origination process, access checks are made by the system on the following:

- Class of Service of an individual terminal
- Trunk Group Access Restrictions (TGAR) code of a terminal, if a direct trunk access code is dialed or as an optional feature when a Basic Alternate Route Selection (BARS) or Network Alternate Route Selection (NARS) access code is dialed
- area code and exchange code, if dialed by terminals with toll denied or conditional toll denied Class of Service, using direct trunk access codes and Code Restriction Tables
- Network Class of Service (NCOS) of a terminal, if Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) or Coordinated Dialing Plan (CDP) access codes are dialed, or if direct trunk access codes are dialed and New Flexible Code Restriction (NCFR) tables are programmed

Previously restricted connections by any feature other than Trunk Barring cannot be lifted or avoided by the Flexible Trunk to Trunk Connections feature. Basically, all existing restrictions apply with the exception of Trunk Barring restrictions.

Call Transfer

If Flexible Trunk to Trunk Connections allows a telephone set to transfer to an outgoing trunk, Access Restrictions can still block the transfer. If a telephone is denied transfer by the Flexible Trunk to Trunk Connections feature, then the transfer is blocked regardless of Access Restrictions.

For a transfer to be completed, both Access Restrictions and Flexible Trunk to Trunk Connections must allow the transfer.

Conference

If the Flexible Trunk to Trunk Connections feature allows a telephone set to conference to an outgoing trunk, then Conference is allowed unless it is blocked by other existing restrictions. If a telephone set disconnects from a conference, Flexible Trunk to Trunk Connections restrictions verify whether the telephone set is allowed to transfer the call between the two trunks. If allowed, this unsupervised conference is completed, unless and until barred by another feature.

Attendant Console Operations

Flexible Trunk to Trunk Connections does not support Attendant Console operations. If an attendant attempts to extend an originating trunk connection on a route barred by the Trunk Barring feature, overflow tone is provided. The Flexible Trunk to Trunk Connections feature does not lift this restriction.

Although Attendant Consoles have a Conference key, Flexible Trunk to Trunk Connections does not apply any restrictions.

Basic Alternate Route Selection
Network Alternate Route Selection
Coordinated Dialing Plan
Flexible Numbering Plan

Regardless of the method of dialing used to originate the call with the outgoing trunk, Flexible Trunk to Trunk Connections restrictions apply for Transfer and Conference.

Call Redirection

Call Forward features

When a telephone set performs Call Forward to an external trunk and receives an incoming trunk call, it may result in a trunk to trunk connection. The Flexible Trunk to Trunk Connections Station Class of Service is not applied when forwarding incoming trunk calls to a barred route.

Call Pickup

The new Station's Classes of Service, introduced by the Flexible Trunk to Trunk Connections feature, do not impose any restrictions on Call Pickup.

Meridian Mail Trunk Access Restriction

Flexible Trunk to Trunk Connections limitations do not apply to Meridian Mail Trunk Access Restriction (MTAR). Irrespective of the Station's Class of Service, external calls are prevented from being transferred/conferenced to Meridian Mail.

Multi-Party Operations - Call Join

The functionality of Flexible Trunk to Trunk Connections applies to conferences made by the Call Join operation.

No Hold Conference

When a Meridian 1 proprietary telephone set is established with a trunk call and a No Hold Conference is initiated, Trunk Barring restrictions do not apply, and the conference is completed. However, if the last internal telephone set involved in the No Hold Conference has a Class of Service of Flexible Trunk to Trunk Connections Conditional (FTTC) or Flexible Trunk to Trunk Connections Restricted (FTTR), then the call is disconnected if that telephone set drops out of the conference.

Scheduled Access Restrictions

With the Flexible Trunk to Trunk Connections feature configured, existing restrictions are not avoided. Additional restrictions imposed by Flexible Trunk to Trunk Connections Classes of Service are introduced when Scheduled Access Restrictions is configured.

Toll Operator Break In

The Flexible Trunk to Trunk Connections Classes of Service have no impact on Toll Operator Break In.

Trunk Access From Any Station

There is no limitation with the new Flexible Trunk to Trunk Connections Station Classes of Service that can restrict the station from picking up the call by Trunk Access From Any Station (TAFAS).

Trunk to Trunk Connection

Flexible Trunk to Trunk Connections takes precedence over the Trunk to Trunk Connection feature.

Virtual Network Services

Flexible Trunk to Trunk Connections does not apply any restrictions to existing Virtual Network Services (VNS) functionality.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Configure Flexible Trunk to Trunk Connections options.
2. LD 10 – Configure Flexible Trunk to Trunk Connections for analog (500/2500 type) sets.
3. LD 11 – Configure Flexible Trunk to Trunk Connections for Meridian 1 proprietary sets.

LD 15 – Configure Flexible Trunk to Trunk Connections options.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	NET	Trunk and network options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

FTOP		Flexible Trunk to Trunk Connections Options.
	(FRES)	FTT feature is inactive.
	TBFT	FTT adds new restrictions on connections not barred by TBAR.
	FTTB	FTT lifts TBAR restrictions for routes barred by TBAR. FTT cannot add any new restrictions for non-barred routes.
	FTLY	All set based trunk to trunk connections for Transfer and Conference are controlled by FTT only.
...		

LD 10 – Configure Flexible Trunk to Trunk Connections for analog (500/2500 type) sets.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	500/2500 type telephone set data block.
TN		Terminal Number
	l s c u	For Large Systems
	c u	For Small Systems and Succession 1000 systems
...		
CLS	(FTTC)	Flexible Trunk to Trunk Connections Conditional (default for new sets).
	FTTU	Flexible Trunk to Trunk Connections Unrestricted (default).
	FTTR	Flexible Trunk to Trunk Connections Restricted.
...		

LD 11 – Configure Flexible Trunk to Trunk Connections for Meridian 1 proprietary sets.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	xxxx	Telephone type, where xxxx is: SL1, 2006, 2008, 2009, 2016, 2018, 2212, 2216, 2317, 2616, 3000.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	(FTTC) FTTU FTTR	Flexible Trunk to Trunk Connections Conditional (default for new sets). Flexible Trunk to Trunk Connections Unrestricted (default for existing sets). Flexible Trunk to Trunk Connections Restricted.
...		

Feature operation

No specific operating procedures are required to use this feature.

Flexible Voice/Data Terminal Number

Contents

This section contains information on the following topics:

Feature description 407
Operating parameters 408
Feature interactions 409
Feature packaging 410
Feature implementation 410
Feature operation 412

Feature description

The Flexible Voice/Data Terminal Number feature allows both bearer (B) channels on the M2000 series Meridian 1 proprietary sets to be available for either voice or data calls on a dynamic (per-call) or static basis. This feature has been developed exclusively for VISIT equipment functionality.

With the dynamic capabilities of this feature, a user has access to two simultaneous voice or two simultaneous data connections on the Time Compression Multiplexing (TCM) loop. This capability is practical for users with various desktop multimedia applications, such as VISIT video, that require various combinations of voice and data connections on a per-call basis.

Dynamic voice/data Terminal Numbers (TNs) have two Directory Numbers (DNs) to place and receive calls. The primary voice DN is assigned to key 00 on the telephone. Another key is assigned to the data DN. This key is designated as the data mode key. All data calls are placed and received using this key. Any other secondary DN keys assigned to a dynamic voice/data TN can place and receive voice calls only.

With the static capabilities of this feature, each B-channel on the set is configured as either voice or data. This provides the opportunity for two voice or two data B-channels on the same TCM loop. This configuration doubles the density of the digital line card (XDLC). Since the TN has either a voice or data Class of Service, calls placed from any DN key on the set are either voice or data.

Operating parameters

There are no restrictions against placing data calls on hold.

When a Terminal Number (TN) is in the voice mode, the short hunting feature is terminated when a Data Mode (DTM) key is encountered.

Data calls to a dynamic voice/data TN are not redirected. All TN redirection features such as Call Forward All Calls and Hunting are applicable to voice calls only. If a data call is not presented to the DTM key the call is given reorder tone.

A Data Mode (DTM) key can be assigned to M2000 series proprietary sets with the exception of the M2006 set.

Data Directory Numbers (DNs) for dynamic voice/data TNs cannot have Multiple Appearance DNs (MADNs).

A dynamic voice/data TN can only have one data DN.

No audible progress tones, such as dial tone or ringback, are provided for data calls to or from dynamic TNs. Only Time Compression Multiplexing (TCM) progress messages are sent for data calls. Audible progress tones are provided for voice calls.

If set relocation takes place, upper and lower TNs of a Time Compression Multiplexing (TCM) loop are relocated together. This occurs even if upper and lower TNs were assigned as dynamic or static. A relocated lower TN (0-15) must be in voice mode. Following relocation, both TNs maintain their prior voice or data settings.

When a service change is performed on a dynamic TN in data mode, it is automatically changed to voice mode.

To prevent improper setup, the new Class of Service Flexible Terminal Number Allowed (FLXA) must be specified to assign Data Class of Service to a lower TN or Voice Class of Service to an upper TN.

Feature interactions

Call Forward All Calls
Call Forward, Internal Calls

Voice calls directed to a dynamic voice/data Terminal Number are forwarded, if either of these features are enabled. Data calls, to a dynamic voice/data TN, are not forwarded.

Call Redirection

If a call is not presented to the Data Mode (DTMK) key, the call is given reorder tone.

Call Waiting
Camp-On

These features are not supported on data calls to a dynamic voice/ data TN.

Call Waiting and Camp On are supported for voice calls to dynamic voice/ data TN. However, no tone is inserted during a Camp On attempt if the Terminal Number is in a busy data mode.

Message Waiting Forward Busy
Call Forward Busy

Voice calls directed to a call processing busy dynamic voice/data TN are redirected using Message Waiting Forward Busy or Call Forward Busy provided these features are configured for the TN. Data calls to dynamic voice/data TNs are not redirected.

Voice Call

If a dynamic TN has a single appearance DN key that terminates on a Voice Call (VCC) key, the called party hears a single beep if occupied on another DN. However, if the called party is a dynamic TN in data mode, the DN key lamp flashes. A beep is not provided.

Feature packaging

The Flexible Voice/Data Terminal Number feature is contained in M2000 Digital Sets (DSET) package 88.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Assign the Static Voice Terminal Number.

2 LD 11 – Assign the Static Data Terminal Number.

3 LD 11 – Assign the Dynamic Terminal Number.

LD 11 – Assign the Static Voice Terminal Number.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.
TYPE:	xxxx	Telephone type, where xxxx = 2006, 2008, 2016, 2216 or 2616.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	FLXA VCE	Flexible voice/data allowed.FLXA is only required if voice TN unit is less than 15. (FLXD) = Flexible voice/data denied (default). This Class of Service can only be assigned to 2006, 2008, 2016, 2216 or 2616 sets. When configured to CLS = FLXA, Voice Class of Service (VCE) can be assigned to the upper TN unit (16 - 31).

LD 11 – Assign the Static Data Terminal Number.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.
TYPE:	xxxx	Telephone type, where xxxx = 2006, 2008, 2016, 2216 or 2616.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	FLXA DTA	Flexible voice/data allowed. FLXA is only required if data TN unit is greater than 15. (FLXD) = Flexible voice/data denied (default). This Class of Service can only be assigned to 2006, 2008, 2016, 2216 or 2616 sets. When configured to CLS = FLXA, Data Class of Service (DTA) can be assigned to the lower TN unit (0 -15).

LD 11 – Assign the Dynamic Terminal Number.

Note: Terminal Numbers with Voice Class of Service and Flexible Voice/Data Allowed can become dynamic voice/data Terminal Number by assigning a secondary SCR/SCN key at the DTMK prompt. The DN specified with this key becomes the data DN.

Prompt	Response	Description
REQ:	NEW CHG	New, or Change.

TYPE:	xxxx	Telephone type, where xxxx = 2006, 2008, 2016, 2216 or 2616.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	FLXA VCE	Flexible voice/data allowed. (FLXD) = Flexible voice/data denied. This Class of Service can only be assigned to 2006, 2008, 2016, 2216 or 2616 sets. When configured to CLS = FLXA Voice Class of Service (VCE) can be assigned to the upper TN unit (16 - 31) and Data Class of Service (DTA) can be assigned to the lower TN unit (0 -15). A Single Call Ringing (SCR) key can be designated a Data Mode (DTMK) key.
DTMK	xx	Key assignment for Data Mode Key. This key must be a single appearance SCR/SCN key and cannot be assigned key 00.
- KEY	00 aaa xxxx	Prime Directory Number Key, where aaa = SCR, SCN, MCR or MCN and xxxx = Voice Directory number
- KEY	xx SCR yyyy xx SCN yyyy	Single Call Ringing Single Call Non Ringing Data Mode Key, where xx = key number and yyyy = Data Directory Number.

When call processing switches between voice and data mode on the dynamic Terminal Number, some Class of Service option data is automatically modified. In data mode, the dynamic TN has options Warning Tone Denied (WTD) and Maintenance Telephone Denied (MTD). When switched back to voice mode, the original settings for these options is automatically restored, and the Class of Service is not printed.

Feature operation

No specific operating procedures are required to use this feature.

Forced Camp-On and Priority Override

Contents

This section contains information on the following topics:

Feature description 413
Operating parameters 413
Feature interactions 414
Feature packaging 415
Feature implementation 415
Feature operation 417

Feature description

Forced Camp-On is similar to the regular station-to-station Camp-On, except that it can be done without an internal or external call on hold. When used with Priority Override, the capability is called Enhanced Override.

Priority Override allows an established call to be broken into and another call presented to the desired party. Before Break-In occurs, a warning tone is given to all parties involved in the established call. The set performing the override must have a priority level equal to or higher than the set being overridden.

Operating parameters

Priority Override and Forced Camp-On can operate independently of each other.

All stations involved in an established call being broken into must have Warning Tone Allowed (WTA) Class of Service.

Priority Override and Forced Camp-On cannot be applied to telephones involved in any of the following:

- non-established call
- conference call
- attendant call
- Release Link attendant call
- attendant call through Centralized Attendant Service or a Primary Rate Access/Integrated Services Digital Network trunk
- Automatic Call Distribution (ACD) call
- data call
- parked call
- call waiting call
- held call
- operator Call Back or toll operator Break-In call
- Make Set Busy active, or
- Do Not Disturb active.

External trunks cannot perform priority override. They can be overridden only if they are the undesired party of an established call being broken into.

Feature interactions

Multi-party Operations

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

Override

When Priority Override is activated, it replaces normal override. Once Priority Override has been performed on a set, its Digit Display shows the DN of the overriding set.

Feature packaging

Priority Override/Forced Camp-On (POVR) is packaged under package 186.

Dependencies:

- Flexible Feature Codes (FFC) package 139
- Multi-party Operations (MPO) package 141

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10, LD 11 – Respond to CLS prompt with CPFA to allow Camp-On to another set, or CPFD to deny such Camp-On. Respond to PLEV prompt with a value between 1 and 7, to set the priority level for this set.

2 LD 11 – Respond to the KEY prompt with a key number, followed by EOVR, to define an Enhanced Override key for each Meridian 1 proprietary telephone.

3 LD 14 – Trunks to be involved in such Camp-On override calls must have Warning Tone Allowed (WTA) Class of Service allowed.

4 LD 15 – To select either automatic or manual forced Camp-On for a customer, respond to the AFCO prompt with either YES (automatic) or NO (manual).

5 LD 16 – At the PLEV prompt, specify priority levels for trunk routes.

6 LD 57 – The Enhanced Override flexible feature code must be defined by responding to the EOVR prompt with an appropriate FFC digit sequence to be assigned that function.

The following are additional to the definitions required for telephone set configuration without this feature.

LD 10, LD 11 – Respond to CLS prompt with CPFA to allow Camp-On to another set, or CPFD to deny such Camp-On. Respond to PLEV prompt with a value between 1 and 7, to set the priority level for this set.

Prompt	Response	Description
...		
CLS	(CPFA) CPTD	Forced Camp-On from another set (allowed) denied.
PLEV	0-(2)-7	Priority Level, prompted with Priority Override/Forced Camp-On (POVR) package 186. 2 = set can override sets of level 1 and 2, and can be overridden by sets of level 2-7.

LD 11 – Respond to the KEY prompt with a key number, followed by EOVR, to define an Enhanced Override key for each Meridian 1 proprietary telephone.

Prompt	Response	Description
...		
KEY	xx EVOR	Key number; Enhanced Override.

LD 14 – Trunks to be involved in such Camp-On override calls must have Warning Tone Allowed (WTA) Class of Service allowed.

Prompt	Response	Description
...		
CLS	(WTA) WTD	Warning Tone (allowed) denied.

LD 15 – To select either automatic or manual forced Camp-On for a customer, respond to the AFCO prompt with either YES (automatic) or NO (manual).

Prompt	Response	Description
...		
AFCO	(NO) YES	(Manual) Automatic Forced Camp-On, prompted with Priority Override/Forced Camp-On (POVR) package 186.

LD 16 – At the PLEV prompt, specify priority levels for trunk routes.

Prompt	Response	Description
...		
PLEV	0-(2)-7	Priority Level, prompted with Priority Override/Forced Camp-On (POVR) package 186. 2 = set can override sets of level 1 and 2, and can be overridden by sets of level 2-7

LD 57 – The Enhanced Override flexible feature code must be defined by responding to the EOVR prompt with an appropriate FFC digit sequence to be assigned that function.

Prompt	Response	Description
...		
- EOVR	xxxx	Enhanced Override (manual Forced Camp-On followed by Priority Override).

Feature operation

Forced Camp-On is activated automatically (if Automatic Forced Camp-On is defined), or it can be activated manually using the Enhanced Override (EOVR) key on Meridian 1 proprietary telephones or the Enhanced Override Flexible Feature Code on analog (500/2500 type) telephones. If the EOVR key is pressed again or the Enhanced Override Flexible Feature Code dialed again, Priority Override is activated.

If Forced Camp-On is not equipped, the first depression of the EOVR key, or the first dialing of the Enhanced Override Flexible Feature Code activates Priority Override.

To activate Priority Override, the user of an analog (500/2500 type) telephone dials the Override Flexible Feature Code, while the user of a Meridian 1 proprietary telephone presses the Override key (OVR). Priority Override can also be activated using the Enhanced Override Flexible Feature Code or the Enhanced Override key (EOVR), as described previously.

Forward No Answer Call Waiting Direct Inward Dialing

Contents

This section contains information on the following topics:

Feature description 419
Operating parameters 419
Feature interactions 420
Feature packaging 420
Feature implementation 420
Feature operation 420

Feature description

The Forward No Answer Call Waiting Direct Inward Dialing (FCWD) feature allows a Direct Inward Dialing (DID) call that encounters a busy set with Call Waiting Allowed to be routed to an attendant (or recalled to the night DN during Night Service), if it is not answered within a customer-defined period (between 2-126 seconds). If Return to Same Attendant is equipped, the call is routed to the first available attendant.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Waiting Redirection

With the Call Waiting Redirection feature also enabled, the Call Waiting Redirection feature takes precedence over the FCWD feature. The existing CFNA also takes precedence over the existing Attendant Recall of Call Waiting calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call while the FCWD feature applies an attendant recall timer, the Call Waiting Redirection feature also has precedence over the FCWD timer.

Feature packaging

This feature is packaged under French Type Approval (FRTA), package 197.

Feature implementation

LD 15 – Respond to FCWD prompt with an even-numbered value between 0 and 126 seconds.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	RDR	Call Redirection.
...		
- FCWD	(0)-126	Number of seconds a DID call should wait on a set before being forwarded to the attendant, prompted with French Type Approval (FRTA) package 197. If (0) is chosen, the call is not forwarded to an attendant. Valid entries are even numbers between 1 and 126; odd numbers are rounded down.

Feature operation

No specific operating procedures are required to use this feature.

Generic XFCOT Software Support

Contents

This section contains information on the following topics:

Feature description 421
Operating parameters 422
Feature interactions 422
Feature packaging 425
Feature implementation 425
Feature operation 427

Feature description

The Generic XFCOT is a circuit card developed to meet the North American Transmission Plan, with the following functionalities:

- tone supervision
- battery supervision
- Periodic Pulse Metering (PPM)
- loopstart signaling

This feature provides the choice of Dynamic Pad Switching in the North American Environment for Central Office trunks (LD 97), enhances the trunk-to-trunk connection, and improves the use of disconnect supervision in features like ACD, Meridian Mail, DISA, Call Park, and Camp-On.

With this feature, a disconnect-supervised loopstart Central Office trunk follows normal XFCOT rules for trunk-to-trunk connection and disconnection.

Functionality is provided on the following IPE circuit cards:

- NTCK16AD for PPM/BAT/BTS
- NTCK16BD for BAT/BTS

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Automatic Call Distribution

This feature is used when a large number of incoming calls are to be answered by a designated group of telephone sets. Calls that cannot be answered immediately are put in an Automatic Call Distribution (ACD) queue.

ACD is allowed on disconnect supervised or unsupervised loopstart trunks. If a caller on an unsupervised loopstart trunk disconnects while the call is in an ACD queue, it is detected when the call is answered by an ACD agent.

With this development, caller disconnection is detected by disconnect-supervised loopstart trunk on an XFCOT card and disconnected callers are then dropped from the ACD queue.

Other ACD operations that require a disconnect-supervised COT such as INTERFLOW, NCFW and NITE RAN are now allowed on a disconnect-supervised loopstart trunk on an XFCOT card.

Call Park

Call Park feature allows an attendant or telephone user to place a call in parked state (connected to a parked DN) where it can be retrieved by any attendant console or station set. If the call is not retrieved after a customer-defined time, the call is recalled to the telephone user who parked it.

Call Park is allowed on disconnect-supervised or unsupervised IPE loopstart Central Office trunks. If a caller on an unsupervised loopstart trunk disconnects while the call is in parked state is detected when the parked call is recalled or answered.

Caller disconnection during park state is detected by a disconnect supervised loopstart trunk on an XFCOT card. The disconnected caller is then dropped from the parked DN.

Camp-On

The Camp-On feature allows an attendant to route one additional call to a busy DN so it can be rung when it becomes free. If the busy DN is not free after a customer-defined time, the call is recalled to the attendant.

A call from a loopstart disconnect supervised or unsupervised loopstart trunk can be camped on. If a caller on an unsupervised loopstart trunk disconnects while the call is camped on, it is detected when the call is recalled or answered.

Caller disconnection during Camp-On operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the camped on call is dropped.

Digital Trunk Interface (DTI) Pad Switching

The DTI pad process intervenes when a DTI port is involved in a connection. It is independent from the normal pad and it handles the DTI port side and the other port side.

This process is changed to handle XFCOT units when the North American Transmission Plan is selected as XUT units.

Direct Inward System Access

This feature allows selected external users to access the system switch by dialing a special directory number, and to use some features of the system as an internal station.

A Direct Inward System Access (DISA) call is allowed on a disconnect supervised or unsupervised loopstart trunk. If a caller on an unsupervised loopstart trunk disconnects during a DISA operation, it is detected by a dial time out or when the call is answered.

Caller disconnection during a DISA operation is detected by a disconnect-supervised loopstart trunk on an XFCOT card and the operation can then be ended.

European XFCOT Software Support

This feature supports international IPE trunks with new functionalities such as supervision on loopstart trunk, PPM, and static pad switching.

The Generic XFCOT Software Support is a product improvement of this feature regarding the pad switching, the trunk-to-trunk disconnection, and the use of disconnect supervision for loopstart trunk in some features.

Meridian Mail

The Meridian Mail feature allows a caller to leave a voice mail message for a person unable to be reached. Once the caller is connected to the voice mail-there is a maximum duration allowed for the message after which the call is disconnected.

Meridian Mail is allowed on disconnect supervised or unsupervised loopstart trunks. If a caller on an unsupervised loopstart trunk disconnects while accessing Meridian Mail, the call is disconnected when the connection-time to the mail box exceeds the maximum duration.

Caller disconnection is detected by the disconnect-supervised loopstart trunk on an XFCOT card and the caller is then dropped from the queue for messaging service or from the mail box.

Periodic Clearing

Periodic Clearing is the sending of a periodic signal from the system to a Central Office when an incoming call has been answered but is not in an established state (for instance, ringing, held, parked). The connection is disconnected if the originator goes on-hook.

The Periodic Clearing condition is timed by the disconnect timer (DCTI) to prevent this situation from lasting for an extended time. When the DCTI timer expires the trunk is disconnected.

The Disconnect Timer can be used without having the feature Periodic Clearing configured particularly when the Central Office trunk has no disconnect supervision. It can be disabled by setting the DCTI to 0 in LD 16.

A loopstart trunk can be marked as disconnect supervised. When it has a class of service providing disconnect supervision, in Periodic Clearing condition the trunk is disconnected when the calling station releases the call.

Feature packaging

This feature is packaged under the following packages:

- Intelligent Peripheral Equipment (XPE) package 203
- International Supplementary Features (SUPP) package 131
- Meridian 1 Enhanced Conference, TDS and MFS (XCT0) package 204
- Meridian 1 Superloop Administration (XCT1) package 205 (unrestricted when the XPE package is equipped)

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 97 – Choose the North American Transmission Plan by answering YES to the NATP prompt.

2 LD 14 – Define the Periodic Pulse Metering parameters on a per country basis. This prompt is used to define a disconnect supervised loopstart trunk on an XFCOT.

LD 97 – Choose the North American Transmission Plan by answering YES to the NATP prompt.

Prompt	Response	Description
REQ	aaa	Request (CHG, END, PRT)
TYPE	LOSP	TYPE = LOSP (Loss Plan Tables)
NATP	YES	North American Transmission Plan for Generic XFCOT

LD 14 – Define the Periodic Pulse Metering parameters on a per country basis. This prompt is used to define a disconnect supervised loopstart trunk on an XFCOT.

Busy tone is provided by the PSTN when the far end releases from outgoing and incoming trunks. The tone supervised COT depends on the busy tone frequency and cadence characteristic of the particular country and is configured on a card basis by responding to the BTID prompt.

Prompt	Response	Description
REQ	NEW	New.
TYPE	COT	Central Office Trunk.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XCOT	Type is IPE COT.
CDEN	(8D)	Card density is 8D.
SIGL	LOP	Loop start signaling.
PPID	xx	Where xx is one of the following: 0 – United Kingdom (50 Hz) 1 – France (12 Khz) 2 – France (50 Hz) 3 – Germany, Egypt, Turkey, Venezuela, Indonesia, Finland (16 Khz) 4 – Switzerland, Ireland, Portugal, Italy, Spain, Lebanon, Turkey (12 Khz) 5 – Denmark (12 Khz) 6 – Norway, Belgium (16 Khz) 7 – Holland (50 Hz) 8 – Australia (two different packs) (12Khz/50 Hz) 9-15 – Reserved for future use.

BTID	xx	Enter the country busy tone ID as follows: 0-2 – Reserved for future use 3 – Germany, Ireland 4 – Switzerland 5 – Denmark 6 – Norway, Kuwait, Chile, Venezuela, Indonesia, Thailand, Korea 7 – Holland 8 – Australia, Mexico 9 – Ireland 10 – Taiwan, Brazil, Tortola, Mexico 11 – Singapore 12 – Argentina, Italy 13 – Lebanon, Italy 14 – Turkey 15 – Reserved for future use.
SUPN	YES (NO)	Trunk Supervision required (not required)
STYP	BTS BAT	Busy tone supervision enabled Loop break supervision enabled
CLS	(LOL) SHL (DIP) DTN (P10) P20 P12	Attenuation pads in (out). Digitone signaling (digipulse). Make-break ration for pulse dialing speed.

Feature operation

No specific operating procedures are required to use this feature.

Group Call

Contents

This section contains information on the following topics:

Feature description 429
Operating parameters 431
Feature interactions 431
Feature packaging 434
Feature implementation 434
Feature operation 435

Feature description

Group Call allows a user of a Meridian 1 proprietary telephone to place a call to up to ten Directory Numbers (DNs) simultaneously by activating a Group Call key. The called DNs must have been previously defined as members of a group.

Each customer within the system can have up to 64 groups assigned. Each group has up to 20 group members. Any DN in the system can be assigned as a member of a group, and a DN can be a member of more than one group. For Small Systems and Succession 1000 systems, six members per group are allowed.

Each Multiple Appearance, MCR/MCN DN reduces the number of telephone sets that can be added to a Group Call. For example, if two telephones have the same MCR appearance of a DN, the number of telephones in the Group Call becomes 19. That is, each appearance of a DN counts as one member, up to a maximum of 20, of the Group Call.

> ***Note:*** Multiple Appearance, SCR/SCN DNs count as one member of a Group Call, irrespective of its number of DN appearances.

Groups are defined through Service Change in LD 18. When a group is defined, each member of the group is assigned a member number. If network or conference blocking is encountered, members are assigned priorities for connection to the Group Call in the order of their group member numbers (member 0 has the highest priority). It is recommended that group members be assigned from different network loops to minimize the possibility of network blocking.

The Group Call key is used to originate a Group Call to all members of the group to which the Group Call key is assigned. The Group Call key for a given group can appear on more than one telephone. More than one Group Call key can be assigned to a group, but only one Group Call key can be active for a given group at any time. A telephone with a Group Call key need not be equipped with a Directory Number (DN) that is defined as a group member.

Activation of a Group Call key originates a call to all assigned members of the group. When the first member of the group answers, ringback tone is removed and a speech path is set up between the member and the originator of the call. As subsequent members answer, they are added to the call. The lamp associated with the Group Call key at the originator's telephone flashes until all members of the group have answered the call.

If a Directory Number (DN) is actively engaged in a call and a Group Call is originated for that DN, either the Group Call is camped on or Call Waiting is activated for the DN and a special warning tone is provided. The special warning tone consists of three rapid bursts of tone followed by 10 seconds of silence, then an additional three rapid bursts of tone.

An active Group Call is under complete control of the originator of the call. If the originator goes on hook, the call is completely broken down. Members who are taking part in a Group Call can disconnect from the call at any time, but once disconnected, they cannot be reconnected.

Operating parameters

A Group Call can be originated only from a Meridian 1 proprietary telephone with a Group Call key.

Group Call does not support data calls.

With the Extended Conference TDS (XCT) card (NT8D17), audio interference can occur if many of the call participants are on “older-style” analog (500/2500-type) sets. This problem does not occur when participants are on Meridian Digital Telephones or “newer-style” electronic analog (500/2500-type) sets. Analog (500/2500-type) sets produce more audio noise and degrade the sound quality of the conference as the number of participants exceeds 12-15 members. As the number of participants drops below this threshold, the sound quality of the Group Call returns to normal.

The maximum number of members per group is 20.

For Small Systems and Succession 1000 systems, the maximum number of members per group is six.

The maximum number of groups per customer is 64.

Each group member DN must have a Warning Tone Allowed Class of Service.

Off-premise Extension (OPX) lines cannot be members of a group.

Calls to a DN that is active in a Conference call, or Group Call, are blocked.

Feature interactions

Automatic Line Selection

This feature is not selected for automatic Outgoing Line Selection or Non-Ringing Line Selection. It is selected for Incoming Ringing Line Selection.

Call Forward All Calls

A Group Call to a telephone with Call Forward active is forwarded one step only. The Call Forward number must be a valid DN.

Call Forward/Hunt Override Via Flexible Feature Code

It is not possible to use Call Forward/Hunt Override FFC as a Group Call DN.

Call Pickup

This feature can be used to answer a Group Call if it is activated by a valid telephone in the same Call Pickup group, or by using Directory Number (DN) Pickup or Group Pickup.

Call Pickup Network Wide

The Group Call feature does not allow a remote party in a Group Call list. Therefore, a Group Call cannot be picked up by a remote station. If during the network scanning a Group Call is found, it will be ignored and the network scanning will continue.

Call Transfer
Conference

Neither Call Transfer nor Conference can be initiated during a Group Call. If an analog (500/2500 type) telephone user flashes the switchhook during an established Group Call, the user is dropped from the call.

Directory Number Delayed Ringing

When a group call is made to an SCN/MCN key with Directory Number Delayed Ringing (DNDR) defined, audible notification will be given after the DNDR delay has expired.

Display of Calling Party Denied

The calling party's display shows the DN of the last set to connect into the Group Call regardless of the Class of Service. The called set displays the Group Number only.

Hold

Only the originator of a Group Call can put the Group Call on hold.

Hot Line

Hot Lines can be members of a Group Call. They cannot, however, have a Group Call key.

ISDN QSIG/EuroISDN Call Completion

Call Completion cannot be applied to a Group Call.

Make Set Busy
Individual Do Not Disturb

A Group Call to a telephone in Make Set Busy or Individual Do Not Disturb mode cannot be completed. The telephone will not be rung and is not counted as part of the Group Call (for instance, if all other members in the group have answered, the lamp next to the Group Call key on the originator's telephone lights steadily).

Network Intercom

When Directory Number Delayed Ringing (DNDR) is defined and an incoming call to set configured with Hot Type I or D Key and DNDR occurs, the set winks until the DNDR timer expires. After this timer expires, the set rings as normal.

Short Buzz for Digital Telephones

The special three-second buzz for Group Call is not affected by this feature.

Telephone features

The following features cannot be applied on a Group Call:

- Call Forward No Answer
- Call Forward Busy
- Call Forward/Hunt Override Via Flexible Feature Code
- Call Join
- Call Park
- Call Transfer
- Conference
- Hunting
- Privacy Release
- Ring Again

Feature packaging

This feature is packaged under Group Call (GRP), package 48 and has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 18 – Add or change a Group Call list.

2 LD 11 – Add or change Group Call for Meridian 1 proprietary telephones.

3 LD 20 – Print Group Call data.

LD 18 – Add or change a Group Call list.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	GRP	Group Call data block.
CUST	xx	Customer number, as defined in LD 15
GRNO	0-63	Number of the Group Call list.
STOR	xx yyy...y <CR>	Group member number (xx) and associated DN (yyy...y). End input of stored Group Call entries

LD 11 – Add or change Group Call for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.

TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx GRC yy	Add a Group Call key, where: xx = key number, and yy = Group Call list number (0-63).

LD 20 – Print Group Call data.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	GRP	Group Call data.
CUST	xx	Customer number, as defined in LD 15
GRNO	0-63 <CR>	Number of the Group Call group. Print data for all Group Call groups.

Feature operation

To make a Group Call:

- Press **Group Call**. All group members are automatically called. The LCD indicator beside the Group Call key flashes until all members have answered. Then it lights steadily.

To make a Group Call using a Flexible Feature Code, see the feature module “Dial Access to Group Calls” on page 81 of this document.

Group Hunt

Contents

This section contains information on the following topics:

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Feature description

Group Hunting is similar to the Hunting feature. If a call encounters a busy DN and a Group Hunting Pilot DN is specified, the call is routed to the next idle DN in a prearranged group. Unlike the existing Hunting feature, Group Hunting allows a customer to:

- Configure all members of a hunt group in one block instead of many different station data blocks.
- Prevent Group Hunt termination on any idle member through a Group Hunt Deactivate Flexible Feature Code (FFC) or through a GHD (Group Hunt Deactivate) key.
- Limit the hunting steps to the total number of DNs in the list.
- Initiate hunting by dialing or accessing a Group Hunt Pilot DN directly.
- Configure a DN to be a member of more than one hunt group.

Pilot DN

Pilot DNs are defined as PLDN Flexible Feature Codes (FFC) in LD 57. Pilot DNs are used in two ways:

1 If the USE prompt is set to GPHT, then the Pilot DN is defined to activate Group Hunting.

2 If the USE prompt is set to Speed Call List Controller (SCLC) or Speed Call List User (SCLU), then the Pilot DN is defined to access the Speed Call or System Speed Call lists that are associated with the Pilot DN.

Termination conditions

When a Group Hunt Pilot DN is dialed, Group Hunting searches the list associated with the Pilot DN, according to the hunt type specified, until one of the following conditions is met:

1 idle DN is encountered

2 Automatic Call Distribution (ACD) DN,
Integrated Voice Messaging Service (VMS) DN,
Message Center (MC) DN,
Listed Directory Number (LDN),
or attendant DN is encountered

3 route access code is encountered

4 ESN access code is encountered

5 Group Hunt Pilot DN is encountered

6 all DNs in the group are hunted to, or
the maximum number of hunting groups is reached

If condition 1 or 2 is met, then incoming calls are completed to that DN. All DNs listed in condition 2 are associated with a queue. Remember the following when configuring these Group Hunt lists:

- These DNs always appear idle to a hunt cycle, regardless of their actual status. The hunt always redirects to the indicated destination, and never comes back into the Group Hunt list, therefore these calls are never queued against the Pilot DN.

- It is recommended that if these DNs must be used in a Group Hunt list, only one such DN be used. This DN must always be the last entry in the list.
- Also, linear hunting must be used. In this configuration, any redirected call is subject to the call processing treatment of the destination.
- Listed DNs can be configured as a last entry in a hunt group list, if linear hunting is used. The redirected call is presented to the associated LDN Incoming Call Indicator (ICI) key on the Attendant Console. The call can be transferred back to the Hunt Group Pilot DN; once transferred, it cannot be recalled to the attendant.
- Attendant DNs can be configured as a last entry in a hunt group list, if linear hunting is used. The call can be transferred back to the Hunt Group Pilot DN; once transferred, it cannot be recalled to the attendant.
- Automatic Call Distribution (ACD) DNs can be configured as a last entry in a hunt group list, if linear hunting is used. The call can be transferred back to the Hunt Group Pilot DN. If the ACD queue has the Hunt Group Pilot DN defined as the night DN, the call is transferred back into the hunt group list.

If termination condition 3 or 4 is met, that call termination depends on either the access code or the number that followed. Therefore, remember the following when configuring Group Hunt lists:

- Use only one access code for each Group Hunt list. The access code must always be the last entry in the list.
- Use linear hunting. In this configuration, any redirected call is subject to the call processing treatment of the destination.
- If an access code is used as a Group Hunt member, it must be entered as “access code and complete destination number” to ensure proper routing to the destination, not just the access code alone.
- Trunk optimization does not apply.

If termination condition 5 is met, the search ends for the current list and begins for the list associated with the new Pilot DN. A Pilot DN cannot be a member of its own Hunting Group.

If termination condition 6 is met, then incoming calls are placed in a queue in the order of arrival. They are then presented to the next DNs in the group as the members become available.

Direct Inward Dialing (DID) calls are placed in a Group Hunting queue only if the group is still in service. If the group is not in service (if all of its members have deactivated Group Hunting), DID calls are routed directly to the attendant.

Calls are removed from a Group Hunting queue when they are abandoned, when they are presented to an available member, or when they are attendant-extended calls and the slow answer recall timer has expired.

Ringback tone is heard by callers who wait in Group Hunting queues for service.

If the attempted DN for termination by Group Hunting is not a valid member or number, an error message (ERR 8985) prints, hunting terminates, and calls route to overflow tone as specified by the intercept treatment.

Hunt types

Two types of Group Hunt are provided: linear and round robin. Only one hunt type is allowed per Group Hunt List.

- Linear: Hunting starts at the first DN in the list and ends when one of the conditions mentioned in “Termination conditions” on page 438 is met.
- Round Robin: Hunting starts at the DN next in the list to the last DN that was hunted to. Hunting ends when one of the conditions mentioned in “Termination conditions” on page 438 is met.

Group Hunt Lists

Group Hunt lists are defined and modified in LD 18. The Pilot DN entered for each list must have been previously defined as a Group Hunt FFC in LD 57. When a Group Hunt list is defined, the members are assigned a member number as in configuring a Speed Call List. The maximum DN size of each member is 31 digits. The list members can have one of the following DN types:

1 Single or Multiple Appearance DN

2 Listed Directory Number

3 Attendant DN

4 Automatic Call Distribution (ACD-DN), VMS-DN, MC-DN

5 Route access code (route access code + number)

6 Electronic Switch Network (ESN) number (for example, access code + number)

7 Group Hunt Pilot DN

8 Radio Paging access code followed by a complete DN

Note: A Group Hunt list can also be modified through a Speed Call or System Speed Call Controller key, through an analog (500/2500 type) telephone feature Speed Call Controller, or through Group Hunting Speed Call or System Speed Call Controller Flexible Feature Codes (FFC).

Composition of Group Hunt lists

Authorized Group Hunt list members belong to one of the following categories:

- Set-associated DNs (DN type 1) — These DNs are associated with sets and/or keys in a stand-alone system. They are any of the following:
  - — Single-appearance DNs
  - — Multiple-appearance, single-call arrangement DNs
  - — Multiple-appearance, multiple-call arrangement DNs

  - If MADNs with multiple-call arrangement are to be used in Group Hunt lists, they must have only one Prime DN appearance.
  - MADNs on analog (500/2500-type) sets with the MCRA class of service (Multiple-call arrangement) are not supported in Group Hunt lists. If one appearance of this MADN is busy, all other appearances are also considered busy by the Group Hunt Cycle.
  - A set-associated DN can only be defined 96 times as a Group Hunt list member in the system.

- System-associated DNs (DN types 2, 3, 4, 7, 8) — These DNs are associated with other destinations than extensions in a stand-alone system.
- Routing-associated DNs (DN types 5, 6) — These DNs are associated with destinations outside the stand-alone system.

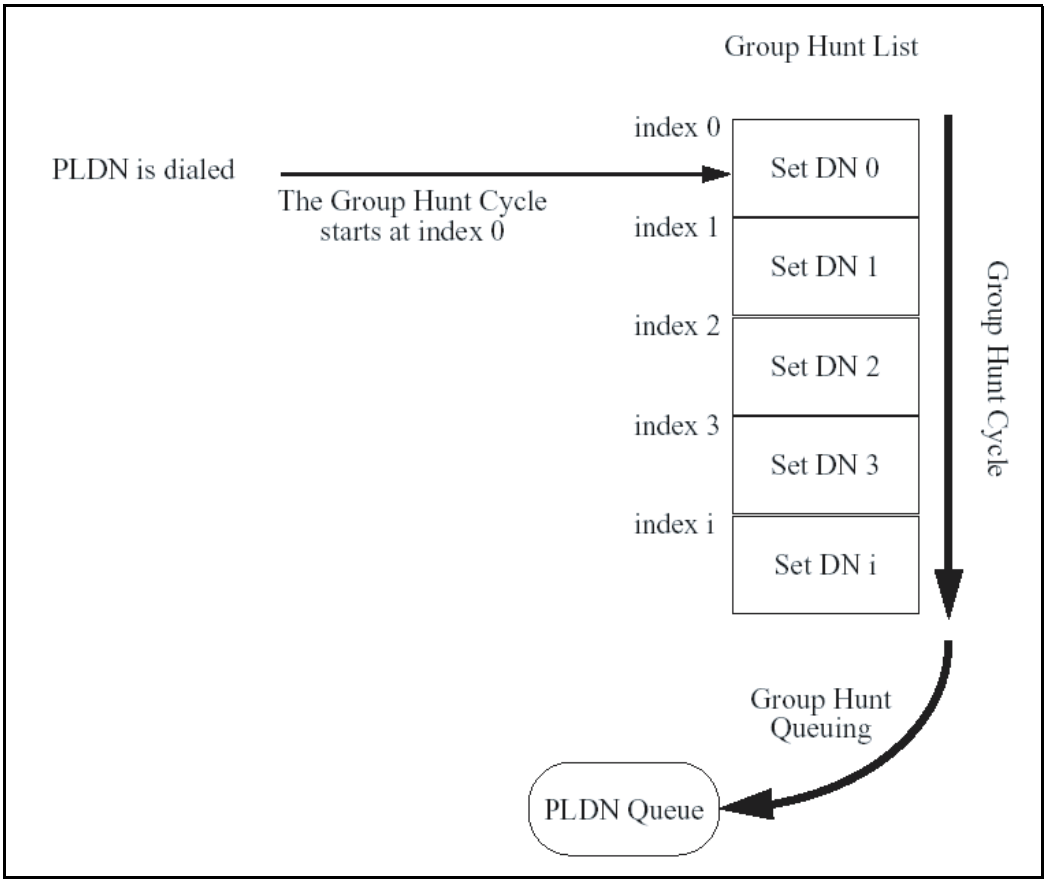
System- or routing-associated DNs always appear available during a Group Hunt cycle. Set-associated DNs do not appear available. Therefore, the Group Hunt cycle always redirects to system-associated and routing-associated DNs when they are met in the list, and the call is never queued against the Pilot DN.

If Linear Hunt is configured for a Group Hunt list that has system- or routing-associated DNs, make these DNs the last entries in the list. List members entered with a higher index are never reached because the system- or routing-associated DNs always route the call.

There are four supported Group Hunt lists:

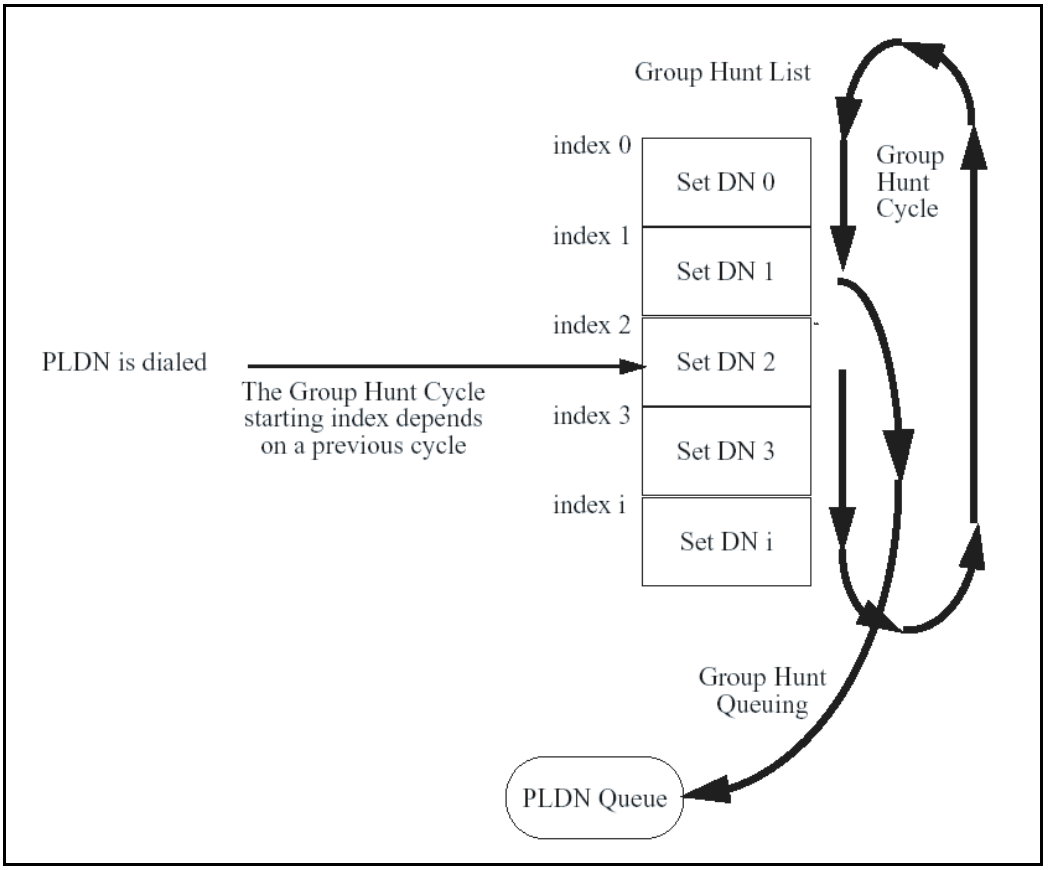
- Type I Group Hunt lists set-associated DNs — These use Linear Hunt type. When the user dials the PLDN, the Group Hunt Cycle offers the call to the first list member. If this member is not eligible, the next member is tried. This process continues until one is found, or until the complete list is searched. Figure 1 on page 443 describes the Group Hunt cycle for Type I Group Hunt lists.

Figure 1
Group Hunt cycle for Type 1 Group Hunt lists



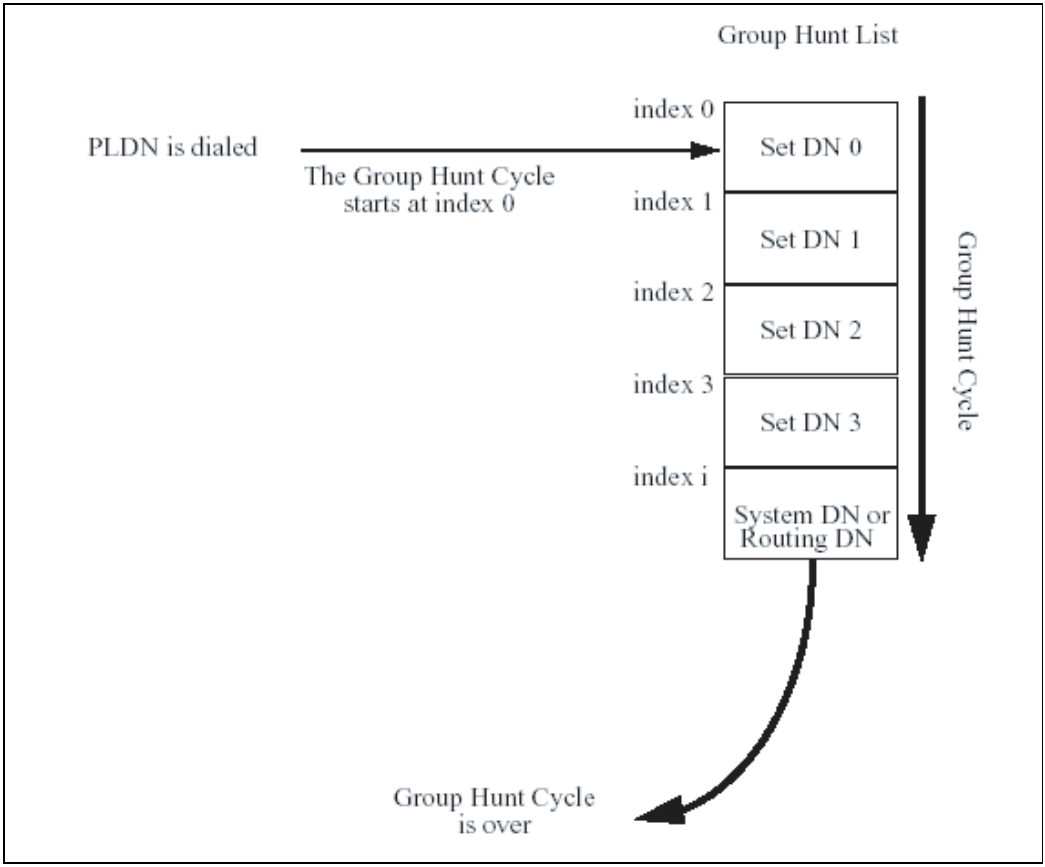
- Type II Group Hunt lists set-associated DNs — These use Round Robin Hunt type. When the user dials the PLDN, the Group Hunt Cycle offers the call to the index next to the last index that was hunted to in the previous cycle. If this member is not eligible, the next DN member is tried. This process continues until one is found, or until the whole list is searched. Figure 2 on page 444 describes the Group Hunt cycle for Type II Group Hunt lists.

Figure 2
Group Hunt cycle for Type II Group Hunt lists



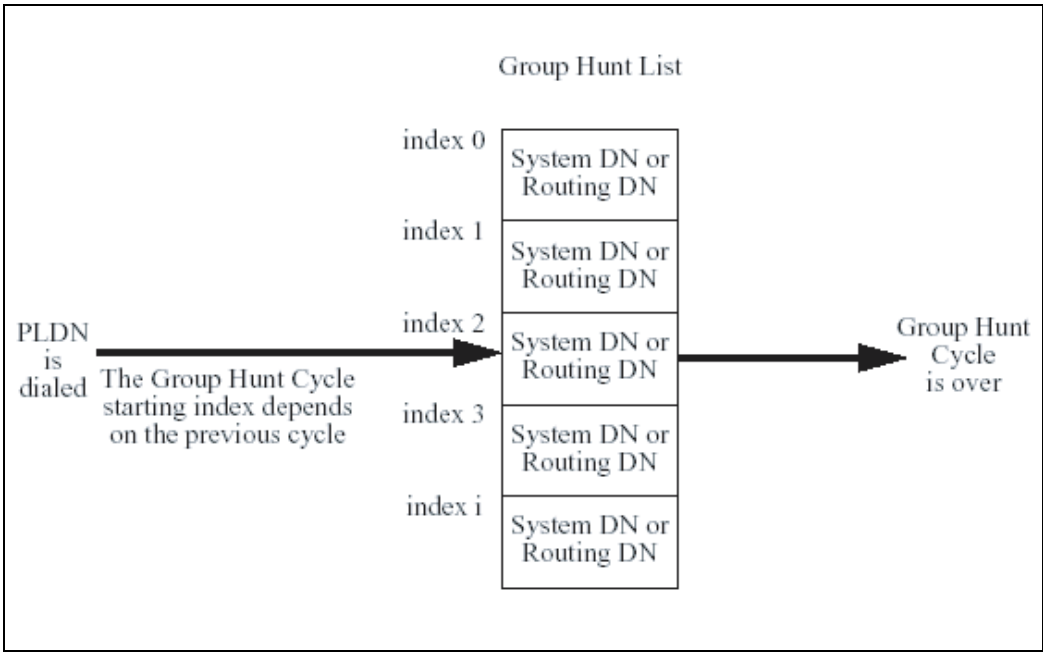
- Type III Group Hunt lists set-associated DNs except for the highest index position, which is filled with a system- or routing-associated DN — These use Linear hunt type. When the user dials the PLDN, the Group Hunt Cycle offers the call to the first list member. If this member is not eligible, the next member is tried. This process continues until one is found. If no member is found, the call is offered to the highest index position (System- or Routing-associated DN). There is no queuing. Figure 3 on page 445 describes the Group Hunt cycle for Type III Group Hunt lists.

Figure 3
Group Hunt cycle for Type III Group Hunt lists



- Type IV Group Hunt lists system- or routing-associated DNs of the same DN type — These use Round Robin hunt type. Once the PLDN has been dialed, the Group Hunt Cycle offers the call to the index next to the last index that was hunted to during the previous cycle. There is no further search in the list nor is there any queuing. Figure 4 on page 446 describes the Group Hunt cycle for Type IV Group Hunt lists.

Figure 4
Group Hunt cycle for Type IV Group Hunt lists



Note: The user must define the Group Hunt list to match one of the four list types.

Queuing

If all members of a Group Hunt list are busy, calls are queued against the Pilot DN of that Group Hunt list. Ring-back tone is provided. There are several options available to control the number of calls allowed to be queued against any given Pilot DN. These options are:

- Group Hunt Queuing Limitation allows the system administrator to select, through service change, the number of calls allowed to queue against the Pilot DN. The selection is made by responding to the Maximum Queue (MQUE) prompt in LD 57. The valid responses to this prompt are:

  — **0**: No calls allowed to queue.

— **1**: One call allowed to queue.

— **ALL**: No limit to the number of calls allowed to queue.

— **ACTM**: The number of calls allowed to queue must be less than or equal to the number of active Group Hunt list members.

Group Hunt Deactivate and Activate

Group Hunt Deactivate allows an idle Set-associated DN to be made non-eligible to Group Hunt calls. A Group Hunt Deactivate FFC code (GHTD) is available to both analog (500/2500-type) and digital sets. The Group Hunt Deactivate key (GHD) can be configured on digital sets.

1 Digital sets

a If the station user activates the DN key, dials the GHTD code and then dials PLDN, the DN is deactivated from the Group Hunt list associated with PLDN. An overflow tone is given if the operation is not successful.

b If the station user activates the DN key and dials the GHTD code, the DN is deactivated from all Group Hunt lists to which this DN belongs. An overflow tone is given if the operation is not successful.

c If the station user activates the GHD key (key lamp is dark), the Prime DN (defined on key 0) is deactivated from all Group Hunt lists to which this Prime DN belongs. The GHD key lamp is lit if the operation is successful. The GHTD code must be used to deactivate non-Prime DNs.

2 Analog (500/2500-type) sets

a If the station user goes offhook, dials the GHTD code, and then dials PLDN, the DN is deactivated from the Group Hunt list associated with PLDN. An overflow tone is given if the operation is not successful.

b If the station user goes offhook and dials the GHTD code, the DN is deactivated from all Group Hunt lists to which this DN belongs. An overflow tone is given if the operation is not successful.

If the DN is a Multiple-Appearance DN, all appearances of the DN are deactivated when a DN member executes one of the deactivation processes.

Group Hunt Activate allows an idle Set-associated DN to return to the active state from the Group Hunt Deactivate state. A Group Hunt Activate FFC code (GHTA) is available to both analog (500/2500-type) and digital sets. A Group Hunt Deactivate key (GHD) can be configured on digital sets.

1 Digital sets

a If the station user activates the DN key, dials the GHTA code and then dials PLDN, the DN is activated again in the Group Hunt list associated with PLDN. Overflow tone is given if the operation is not successful.

b If the station user activates the DN key and dials the GHTA code, the DN is activated again for all Group Hunt lists to which this DN belongs. Overflow tone is given if the operation is not successful.

c If the station user activates the GHD key (key lamp is lit), the Prime DN (defined on key 0) is activated again for all Group Hunt lists to which this Prime DN belongs. The GHD key lamp is darkened if the operation is successful.

2 Analog (500/2500-type) sets

a If the user goes offhook and dials the GHTA code, and then dials PLDN, the DN is activated again for the Group Hunt list associated with PLDN. An overflow tone is given if the operation is not successful.

b If the station user goes offhook and dials the GHTA code, the DN is activated again for all Group Hunt lists to which this DN belongs. An overflow tone is given if the operation is not successful. The GHTA code must be used to activate non-Prime DNs.

If the DN is a Multiple-Appearance DN, all appearances of the DN are activated again when a DN member executes an activation processes.

Access to Group Hunt lists

A Group Hunt list can be accessed by dialing the associated Pilot DN, through:

- manual dialing
- automatic dialing (such as Autodial, Hotline, Speed Call)
- redirection (such as Call Transfer, Call Forward, Hunt)
- ACD Night Service
- ACD interflow/overflow
- trunk access

A Pilot DN can be accessed like any other DN in the network. Any network user can access all Group Hunt lists defined for a network from anywhere in the network. This allows a centralized Group Hunt list to be set up for all network users.

However, Group Hunting is not possible across the network because calls encountering access code entries are always directed to the destination and never return to the hunt queue.

Operating parameters

The Group Hunting feature does not support data calls.

Hunting is limited to the following:

- the total number of DNs in the group
- a maximum of 30 hunting groups for each hunting sequence (for multi-group systems)
- a maximum of 18 hunting groups for each hunting sequence (for all other systems)

Hunting can be limited to the total number of DNs in the group, to 30 hunting groups per hunting sequence for multi-group systems, or to 18 hunting groups per hunting sequence for all other systems.

A maximum of 31 digits can be entered in each list entry.

A maximum of 96 entries can be placed in each list.

A specific station can be defined within a group, among different groups, or a combination thereof a maximum of 96 times.

A maximum of 8000 Group Hunt lists can be defined on a system (programmable through the existing MSCL prompt in LD 17 and reduced by the number of defined Speed Call and System Speed Call lists.)

For larger applications, the ACD package must be equipped to optimize call control and call distribution.

It is recommended that the Group Hunt feature be primarily used with set-associated DNs.

A Group Hunt pilot DN cannot be a member of its own list.

Round Robin hunting should only be used if all entries in the Group Hunt list are the same type (for example, all set-associated DNs or system-associated DNs).

A Pilot DN can be accessed from a network TIE trunk. Also, members of the Group Hunt list can be located at remote nodes.

Feature interactions

Attendant Timed Recalls

Attendant-extended calls to a PLDN will recall to the attendant when the recall timer expires.

- If the call extends to an eligible member in the Group Hunt list, the Slow Answer Recall Timer of the customer applies.
- If the attendant-extended call is queued to the PLDN, the Call Waiting Timer of the customer applies.
- If the attendant-extended call is queued to the PLDN, the Call Waiting Recall Timer is started as explained above; when a list member becomes eligible and is offered the call, the timer re-starts with its Slow Answer Recall value.

If the call is extended to an eligible member that has Call Forward No Answer configured, the Flexible Call Forward No Answer Timer applies instead of the Slow Answer Recall Timer. Ringing in the Group Hunt list continues as long as allowed by CFNA.

Access Restrictions

If a routing-associated DN is programmed in a Group Hunt list, access restrictions apply, based on the Class of Service of the calling station or route, the TGAR of the calling station or route, or both.

Attendant Alternative Answering

A Pilot DN can be defined as an alternative DN. Calls forwarded to a Pilot DN as an alternative DN are directed to the next DN in the group.

Attendant Blocking of Directory Number

It is not possible to activate the Attendant Blocking of DN feature for a Pilot DN. If an attempt is made to block a PLDN, the attempt will be canceled and overflow tone will be returned. If a DN that is a member in a Group Hunt (or Hunt) list is blocked by the Attendant Blocking of DN feature, the DN is considered to be busy.

Attendant Break-in and Toll Operator Break-in

Attendant Break-in and Toll Operator Break-in will not be supported when dialing a Pilot DN directly.

Attendant Busy Verify

An attendant is not allowed to busy-verify when dialing a Pilot DN directly.

Attendant Overflow Position

A PLDN cannot be configured as an Attendant Overflow DN (AODN).

Call Forward All Calls

When Group Hunt attempts to terminate on a DN, which has Call Forward All Calls active, it will continue with the next DN in the group if the attempted DN is busy, or if the DN is idle and the response to the Call Forward Ignore (CFWI) prompt in LD 57 is NO. If the attempted DN is idle and the response to the CFWI prompt in LD 57 is YES, then the system terminates Group Hunt and ring stations associated with the DN.

Call Forward Busy

Group Hunting has priority over the Call Forward Busy feature.

If the DN to be terminate has FBA (Forward Busy Allowed) Class of Service is busy, then Group Hunting continues with the next DN in the group.

Call Forward by Call Type

A Pilot DN can be configured as the redirection DN (HUNT, FDN, EHT, EFD) for the CFCT feature. The interaction is the same as for the Call Forward No Answer feature.

Call Forward External Deny

A Pilot DN cannot be configured as the Call Forward All Calls redirection DN if the set has the CFXD capability allowed.

Call Forward/Hunt Override through Flexible Feature Code

Primary Line Directory Numbers (PLDNs) are not overridden by the Call Forward/Hunt Override Via FFC feature. Any attempt will be ignored and access denied treatment will result.

Call Forward No Answer

Call Forward No Answer (CFNA) can optionally be configured to use a Pilot DN. This option is available when the HUNT DN or the FDN is defined as a Pilot DN.

If an idle station attempted for termination has CFNA defined, then the station will be rung. If the station does not answer within the customer specified number of ring cycles, then Group Hunting will continue with the next DN in the group. The calling party will continue to hear ringback tone until one of the conditions mentioned in “Termination conditions” on page 438 (the last condition is not applicable in this case) is met, or until they release the call.

Call Forward No Answer, Second Level

Second Level Call Forward No Answer will not be applied to calls that are using Group Hunt.

Call Forward No Answer by Call Type

CFNA by Call Type can be configured to use a Pilot DN. This option is available when the EFD or EHT DN is defined as a Pilot DN.

When Group Hunting terminates on an idle station with Call Forward No Answer by Call Type active, treatment will be the same as in the case of CFNA.

Call Forward No Answer, Second Level

Second Level CFNA will not be applied to calls with Group Hunting active.

Call Detail Recording on Redirected Incoming Calls

For the Call Detail Recording on Redirected Incoming Calls feature, in the case of Group Hunt, the Pilot DN is the one before the last set in the redirection chain.

Call Transfer

Any call can be transferred to a Group Hunt Pilot DN. If there are no idle sets available for the call transfer, the call is queued to the Pilot DN and the caller receives ringback tone. If the call cannot be queued because the queue threshold has been reached, the caller receives busy tone.

Call Waiting

Call Waiting to a Pilot DN is not be supported.

Camp-on

Camping an incoming call on to a Pilot DN is not be supported.

Digit Display and Name Display

Until a call is answered, the calling party sees the dialed DN. When the call is answered, the caller sees the dialed DN appended with the DN and name of the calling party, if Calling Party Name Display (CPND) is equipped. The terminating set will always see the originating DN appended with a Pilot DN.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

Only basic DPNSS1 UDP calls are supported with Group Hunting. Interactions between DPNSS1 Supplementary Services and Group Hunting are not supported.

DPNSS1 Diversion

Only simple DPNSS1 calls support Group Hunting. All DPNSS1 supplementary services do not support Group Hunting.

Do Not Disturb

Do Not Disturb (DND) has priority over Group Hunting. Group Hunting will skip over sets with DND active.

Enhanced Night Service

If a Pilot DN is defined as one of the NITE DNs from the list associated with the Trunk Night Group, then incoming calls directed to the Pilot DN will be presented to the next idle DN in the hunt group.

Electronic Switched Network

Group Hunting can be applied to Network calls. An Electronic Switched Network (ESN) access code (trunk steering code), if encountered during Group Hunting, will terminate the hunting sequence.

Hunting

Group Hunting has priority over Hunting. If the DN attempted for termination by Group Hunting has HTA COS, and if it is busy, Group Hunting continues with the next DN in the group instead of following the DN's hunting configuration.

ISDN QSIG/EuroISDN Call Completion

Call Completion to Busy Subscriber cannot be applied to Pilot DN when no idle set is located during a Group Hunt call.

Last Number Redial
Stored Number Redial

A Pilot DN will be stored as a Last Number Redial (LNR) and Stored Number Redial (SNR) number when it is dialed directly.

Make Set Busy

Make Set Busy (MSB) has priority over Group Hunting. Group Hunting will skip over sets with MSB active.

Multiple Appearance Directory Number

While Multiple Appearance DNs (MADN) single call arrangements are treated the same as Single Appearance DNs (SADN), MADN multiple call arrangements must be avoided in a Group Hunt list.

With MADN multiple call arrangement, the idle or busy status of the MADN is determined by the Terminal Number (TN) data block of the prime appearance of the called DN. If there is more than one prime appearance of the called DN, the idle or busy status is then selected from the last TN in the DN block for the MADN (DNB prompt in LD 22). This means that there can be idle appearances of the MADN, while the hunt cycle regards them as busy and attempts to terminate on the next idle member of the Group Hunt list.

If an MADN multiple call arrangement must be used, a supervisor set must be assigned to the hunt group. This supervisor set must be given the one and only prime appearance of the MADN. Any other appearance must have the MADN programmed as a secondary DN (any DN key other than 0). In this way, the supervisor set controls the status of the MADN and thus the Group Hunt treatment. If the supervisor set is busy, the hunt does not terminate on the MADN.

Multi-Party Operations

As per the existing Multi-Party Operations (MPO) feature, recovery of misoperation of Call Transfer will not be applied to incoming calls which are transferred on ringing to a Pilot DN by transferring parties who are waiting in GPHT queues for service.

Night Answer by Time of Day

If a Pilot DN is defined as one of the NITE DNs in LD 15, then incoming calls directed to the Pilot DN will be presented to the next idle DN in the group. At the instant of changeover (change from one night DN to another), Group Hunting, if still active, will keep on hunting for the next idle DN in the group.

Night Service

If a Pilot DN is defined as a NITE DN or trunk NITE DN, then incoming calls directed to the NITE DN or trunk NITE DN will be presented to the next idle station in the hunt group.

On Hold on Loudspeaker

Group Hunt to a loudspeaker DN can be programmed, but will be ignored if configured as Make Set Busy (MSB) by call processing.

Override
Ring Again

Override and Ring Again will not be supported.

Recall to Same Attendant

Calls redirected from a Group Hunt list through the listed DN or flexible attendant DN, and transferred back to the Pilot DN, are recalled if the Slow Answer Recall Timer expires. However, in practical configurations, the hunt terminates on the entry with the listed DN or attendant DN before the Slow Answer Recall Timer expires; consequently, the call is not redirected to that DN and presented on the applicable ICI key on the console. Therefore, the call is never presented as a recall, so that Recall to the Same Attendant does not apply.

Recorded Announcement

Calls which are queued against the Group Hunt Pilot DN cannot receive Recorded Announcement.

Remote Call Forward

If Call Forward All Calls is activated remotely, the interaction with Group Hunting is the same as Call Forward All Calls.

Ring Again on No Answer

Ring Again on No Answer cannot be applied if the DN dialed was a Pilot DN.

Slow Answer Recall

Calls extended by the attendant to the Group Hunt Pilot DN are recalled to the same attendant, after the Slow Answer Recall timer expires. This only applies to a standalone configuration; Network Attendant Service (NAS) is not supported.

Tenant Service

If a Pilot DN is defined as a Tenant NITE DN, then incoming calls directed to the Pilot DN will be presented to the next idle DN in the hunt group.

Total Redirection Count

Group Hunt takes precedence over the Total Redirection Count feature, in that the TRCNT limit is not applied to a Group Hunt call.

Warning Tone

Warning Tone is not applied to queued calls, if the French Type Approval package (197) is not equipped. If the French Type Approval package (197) is equipped, a warning tone of Camp-on can be provided to the first active member of a Group Hunt list that has Warning Tone Allowed (WTA) Class of Service (COS). Any new call in the queue is announced to the next set in the hunt chain that has WTA COS.

16-Button Digitone/Multifrequency Operation

Group Hunt Pilot DN (GRHP) function will not be supported. Group Hunting and Speed Call DN Access can be accessed through the Autodial function.

Feature packaging

Group Hunt requires the following packages:

For markets other than France:

- Group Hunt/DN Access to SCL (PLDN) package 120, which has the following dependencies:
  - System Speed Call (SSC) package 34
  - International Supplementary Features (SUPP) package 131 where applicable
  - Flexible Feature Codes (FFC) package 139

For the French market only:

- French Type Approval (FRTA) package 197 and Group Hunt/DN Access to SCL (PLDN) package 120, which has the following dependencies:
  - System Speed Call (SSC) package 34
  - International Supplementary Features (SUPP) package 131
  - Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 22 – Verify that software package 120 is equipped.

1 LD 17 – Enter the number of Group Hunt lists allowed in the system.

2 LD 15 – Enter a Group Hunt PLDN.

3 LD 10 – Enter a Group Hunt Pilot DN (PLDN).

4 LD 11 – Enter a Group Hunting Denied (GHD) key and enter a Group Hunt PLDN. LD 11 is modified to disallow the removal of the last appearance of a Single Call Non-ringing (SCN), Single Call Ringing (SCR), Multiple Call Non-ringing (MCN), or Multiple Call Ringing (MCR) DN which is part of a Group Hunt list. This ensures the DN is removed from all Group Hunt lists prior to being removed from a set.

5 LD 12 – Enter a Group Hunt PLDN.

6 LD 14 – Enter a Group Hunt PLDN.

7 LD 18 – Create or modify Group Hunt lists. Responses are required to the following prompts when a Group Hunt list is modified, created, or removed. This overlay disallows the removal of a Group Hunt list if it is still associated with a PLDN that exists in LD 57. This ensures the PLDN is removed prior to removing the Group Hunt list.

8 LD 20 – Print the Group Hunt list. This includes PLDN entries.

9 LD 57 – Define, change, or print data associated with FFC.

10 LD 57 – Configure Flexible Feature Codes data block for Group Hunt Termination.

11 LD 81 – Count or list all stations equipped with the GHD key.

12 LD 83 – Display the GHD key data.

LD 22 – Verify that software package 120 is equipped.

Prompt	Response	Description
REQ	PRT	Print information.
TYPE	PKG	Type of information to print: equipped packages.

LD 17 – Enter the number of Group Hunt lists allowed in the system.

Prompt	Response	Description
REQ:	CHG	Change the Configuration Record.
TYPE:	PARM	System Parameters
...	...	
MSCL	0-8191	Number of Group Hunt lists allowed in the system.

LD 15 – Enter a Group Hunt PLDN. (Part 1 of 2)

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	NIT	Night Service Options
...		
- NIT1	x...x	First Night service by time of day DN **can be defined as a PLDN.**

LD 15 – Enter a Group Hunt PLDN. (Part 2 of 2)

- TIM1	...	Hour and Minute for First Night Service DN.
- NIT2	x...x	Second Night service by time of day DN **can be defined as a PLDN.**
- TIM2	...	Time for Second Night Service DN.
- NIT3	x...x	Third Night service by time of day DN **can be defined as a PLDN.**
- TIM3	...	Time for Third Night Service DN.
- NIT4	x...x	Fourth Night service by time of day DN **can be defined as a PLDN.**
- TIM4	...	Hour and Minute for Fourth Night Service DN.

LD 10 – Enter a Group Hunt Pilot DN (PLDN). (Part 1 of 2)

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data, or change existing data.
...		
IAPG	0-15	Meridian Link Unsolicited Status Message (USM) group
HUNT	x...x	Hunt DN of the next station in the Hunt chain. Hunt DN can be defined as a PLDN.
...		
AACD	(NO) YES	Associate set (AST) telephone
FTR	FTR	Enter the feature name and related data.
	EFD x...x	External Flexible call forward DN (a Group Hunt pilot can be entered.) External Call Forward No Answer DN can be defined as a PLDN.

LD 10 – Enter a Group Hunt Pilot DN (PLDN). (Part 2 of 2)

	EHT x...x	External Hunt DN External Hunt DN can be defined as a PLDN.
	FDN x...x	Flexible Call Forward No Answer Call Forward No Answer DN can be defined as a PLDN.

LD 11 – Enter a Group Hunting Denied (GHD) key and enter a Group Hunt PLDN. **(Part 1 of 2)**

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
...		
AOM	0-2	Number of Add-on Modules. AOM appears if TYPE = M2216 and M2616
FDN	x...x	Flexible CFNA DN Call Forward No Answer DN can be defined as a PLDN.
...		
ICT	0-<NIPN>	Intercept Computer Terminal or printer number Number of Intercept Positions (NIPN) is defined in LD 15.
EFD	x...x	Flexible CFNA DN for External calls External Call Forward No Answer DN can be defined as a PLDN.
HUNT	x...x	Hunt DN of next station in hunt chain Hunt DN can be defined as a PLDN.
EHT	x...x	External Hunt DN External Hunt DN can be defined as a PLDN.
...		
LANG	(0)-5 X	Language choice for Automatic Wakeup (AWU) calls. Prompted with Multi-language Wakeup (MLWU) package 206.

LD 11 – Enter a Group Hunting Denied (GHD) key and enter a Group Hunt PLDN. **(Part 2 of 2)**

KEY	xx aaa yyyy	Telephone key assignments.
	xx CFW yy z...z	Call Forward key Key number (xx), Call Forward function (CFW), length (yy), Call Forward target DN (z...z) can be defined as a PLDN.
	xx GHD	Key number (xx), Group Hunting Denied function (GHD). The GHD key is added to allow a station user to toggle the Primary (key 0) Directory Number (PDN) in and out of all groups of which that PDN is a member.

LD 12 – Enter a Group Hunt PLDN.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data, or change existing data.
...		
ICP	(NO) YES	Intercept Computer available.
AADN	x...x	Attendant Alternate Answering DN. Alternate Answering DN can be defined as a PLDN.

LD 14 – Enter a Group Hunt PLDN. **(Part 1 of 2)**

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data, or change existing data
...		
NGRP	(0)-9	Night Service Group number

LD 14 – Enter a Group Hunt PLDN. **(Part 2 of 2)**

NITE	x...x	Night Service directory number Night service DN can be defined as a PLDN.
ATDN	x...x	Auto-terminate DN Auto-terminate DN can be defined as a PLDN.
MNDN	x...x	Manual Directory Number Manual DN can be defined as a PLDN.

LD 18 – Create or modify Group Hunt lists. (Part 1 of 2)

Prompt	**Response**	**Description**
REQ	CHG MOV NEW OUT	Change, move, create, or remove a data block.
TYPE	GHT	Group Hunt data block
LSNO	 1-8190 <CR>	List Number Group Hunt lists Use only when REQ = CHG and TYPE = GHT
CUST	 0-99 0-31	Customer number, as defined in LD 15. For Large Systems For Small Systems and Succession 1000 systems
PLDN	x...x	Pilot DN: Prompted when REQ = NEW or CHG and LSNO = <CR>.
DNSZ	4-(16)-31	Directory Number Size Maximum length of DN allowed for Group Hunt list. Prompted when REQ = NEW or CHG and LSNO = <CR>. After DNSZ is defined, it should not be changed. Print the list in LD 20, remove it with REQ = OUT, and rebuild the list with the new DNSZ.
SIZE	 1-96 1-1000	Size of list Maximum DNs in Group Hunt list Range is 1 to 96 entries if response to TYPE is GHT. Range is 1 to 1000 if response to TYPE is SCL or SSC.

LD 18 – Create or modify Group Hunt lists. (Part 2 of 2)

STOR	...	Store: Enter entry (member) number (x...x) and Group Hunt target DN (y...y). For TYPE = GHT the input format is Group Hunt entry and digits stored against it: Where:
	x...x	x...x = GHT entry number from 0 to 95
	y...y	y...y = digits stored
	<CR>	Stop STOR prompt In Group Hunting the member number must conform with the SIZE; the number of digits must conform with DNSZ.
	xx<space><CR>	Remove entry
WRT	(YES) NO	Write Write information to data store.

LD 20 – Print the Group Hunt list.

Prompt	**Response**	**Description**
REQ	PRT	Print data block.
TYPE	GHT	Type of data block: Group Hunt list.
LSNO	1-8190 <CR>	Group Hunt lists. Print all lists
SIZE 1-96		The list size is printed if a Group Hunt list number is entered against the LSNO prompt. SIZE is not printed if <CR> is entered against the LSNO prompt.

LD 57 – Define, change, or print data associated with FFC. (Part 1 of 2)

Prompt	**Response**	**Description**
REQ	CHG NEW	Add new data, or change existing data.

LD 57 – Define, change, or print data associated with FFC. (Part 2 of 2)

TYPE	FFC	Flexible Feature Codes data block.
CUST	 0-99 0-31	Customer number, as defined in LD 15 For Large Systems For Small Systems and Succession 1000 systems
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Code to be modified or created: Pilot DN.
- PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be modified or created. Enter a carriage return to proceed to next prompt.
-- USE	GPHT	USE: enter USE for Pilot DN Group Hunting.
-- LSNO	xxxx	List Number: enter Group Hunt list number. Group Hunt list must exist in LD 18.
-- HTYP	(LIN) RRB	Hunting Type: enter either (Linear) or Round Robin as the type of hunting to be used for the Group Hunt list.
-- CFWI	(NO) YES	Call Forward All Calls Idle: Where: enter NO if Group Hunting is to skip idle stations with Call Forward All Calls active, or enter YES if Group Hunting is to terminate on idle stations with Call Forward All Calls active.
MQUE	 0 1 (ALL) ACTM	Maximum Queue (maximum number of calls allowed to queue against the Pilot DN. Where: Enter 0 to deny all calls from queuing Enter 1 to allow only one no call to queue Enter ALL, the default, to allow all calls to queue or Enter ACTM to limit the number of calls allowed to queue to be less than or equal to the number of active members of the Group Hunt list.

LD 57 – Configure Flexible Feature Codes data block for Group Hunt Termination.

Prompt	Response	Description
REQ	CHG NEW	Add new data, or change existing data.
TYPE	FFC	Flexible Feature Codes data block.
CUST	 0-99 0-31	Customer number, as defined in LD 15. For Large Systems For Small Systems and Succession 1000 systems
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	GHTA	Code to be modified or created: Group Hunt Termination Allowed.
- GHTA	x...x	Enter code to be dialed to allow Group Hunt termination on a set.
CODE	GHTD	Code to be modified or created: Group Hunt Termination Denied.
- GHTD	x...x	Enter code to be dialed to deny Group Hunt termination on a set.

LD 81 – Count or list all stations equipped with the GHD key. (Part 1 of 2)

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
	CNT	Print a count of telephones equipped with the features specified in response to the FEAT prompt.
	LST	List telephones equipped with the features specified in response to the FEAT prompt.
...	...	

LD 81 – Count or list all stations equipped with the GHD key. (Part 2 of 2)

FEAT	GHD	Group Hunt Deactivation key
FEAT	<CR>	<CR> enters the default

LD 83 – Display the GHD key data.

Prompt	**Response**	**Description**
REQ	TNB	Print the TN blocks in Designation order.
...	...	
... KEY xx GHD ...		GHD key data is printed each time it is configured on a set.

Feature operation

No specific operating procedures are required to use this feature.

Group Hunting Queuing Limitation

Contents

This section contains information on the following topics:

Feature description 469
Operating parameters 470
Feature interactions 470
Feature packaging 471
Feature implementation 471
Feature operation 472

Feature description

The Group Hunting Queuing Limitation feature restricts the maximum number of calls that can be queued against a Pilot Directory Number (DN).

The Group Hunting Queuing Limitation feature adds a prompt (MQUE - Maximum Queue) in LD 57 which allows a user to define a limit on the number of calls queued against a Pilot DN. The valid inputs are:

- 0 No calls can be queued.
- 1 One call can be queued.
- ALL All calls may be queued (default).
- <CR> Setting is left as is.

When the maximum is exceeded, the next call that attempts to queue will be given busy treatment.

The following are examples of the treatment calls receive with MQUE set to the various settings:

MQUE set to 0

1 Pilot DN Z can hunt two sets, A and B. Both of these sets are busy.

2 Set (or DID trunk) C dials Pilot DN Z.

3 If C is a set it receives busy tone and cannot be queued, but if it is a Direct Inward Dialing (DID) trunk it receives whatever busy treatment has been requested for that DID route.

MQUE set to 1

1 Pilot DN Z can hunt two sets, A and B. Both of these sets are busy.

2 Set C dials the Pilot DN Z. The call is queued.

3 Set D dials the Pilot DN Z. This call receives busy tone.

4 Set A goes on-hook first. The first call is presented to set A.

MQUE set to ALL

This option disables the Group Hunt Queuing Limitation enhancement. With ALL selected there is no limit as to the number of calls which can be queued against the Pilot DN.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Camp-on

No Camp-on tone is provided for Group Hunting.

Music

No music is provided for Group Hunting Queuing Limitation.

Feature packaging

This feature is packaged under the International Supplementary Features (SUPP) package 131; Group Hunt/DN Access to SCL (PLDN) package 120; and all PLDN package (120) dependencies.

Feature implementation

LD 57 – Modify, create, or print Flexible Feature Codes. The MQUE prompt accepts a limit for the number of calls allowed to be queued against a Pilot DN. When printing the Flexible Feature Codes (FFC) data block, the value against the MQUE prompt is displayed.

Prompt	Response	Description
REQ	CHG NEW PRT	Request: Modify, create, or print data block.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
...	xxxx	Automatic Call Distribution Directory Number.
CODE	PLDN	Code to modify, create: Pilot Directory Number.
- PLDN	XXXX	Enter PLDN to be modified or created.
- - USE	GPHT	Use of this PLDN, Group Hunt Pilot DN.
...		
- - CFWI	...	
- - MQUE		Maximum Queue – Maximum number of calls that may be queued against a Group Hunt Pilot DN.
	(ALL) 0 1 <cr>	All calls may be queued (default). No calls can be queued. One call may be queued. Use default setting if this is a new Pilot DN, leave existing setting as is if the Pilot DN is being modified.
...		

Printing the FFC data block will include the MQUE prompt and its response.

Feature operation

No specific operating procedures are required to use this feature.

Group Hunting Queuing Limitation Enhancement

Contents

This section contains information on the following topics:

Feature description 473
Operating parameters 474
Feature interactions 474
Feature packaging 474
Feature implementation 475
Feature operation 476

Feature description

This feature introduces a Group Hunt Threshold (GHTH) which limits the number of calls that can be linked in the Pilot DN (PLDN) waiting queue. The threshold is calculated dynamically and is set equal to the number of active members in the group hunt list. This limits the number of calls in the PLDN queue to one per active member of the group hunt list. The feature is activated using the MQUE prompt in LD 57. The MQUE prompt now accepts a response of ACTM (Active Members) to invoke the GHTH.

Along with the Group Hunt Threshold this feature introduces the use of Camp-on tone to indicate that there are queued calls.

Operating parameters

Although Automatic Call Distribution (ACD) DNs, Integrated Voice Messaging Service (VMS) DNs, Listed Directory Numbers (LDNs), Route access codes, Electronic Switched Network (ESN) access codes, and other Pilot DNs can be defined as a group hunt list member, it is recommended that they are not used due to the fact that these targets are considered as active when computing the threshold, regardless of their actual state.

Feature interactions

Call Forward by Call Type
Call Forward No Answer

An external call is made to the PLDN. An idle group hunt list member station is rung but does not answer. If the member station has Call Forward No Answer (FNA) or Call Forward by Call Type Allowed (CFTA) Class of Service, then the call is transferred to the attendant after the number of ring cycles defined for Call Forward No Answer has been reached. If the call is an internal call, then the system searches for another idle group hunt list member.

Call Transfer

If a call is transferred to the PLDN, and all Group Hunt list members are busy, the call is queued to the PLDN, if the number of queued calls is less than the Group Hunt Threshold limit. If the number of queued calls has reached the Group Hunt Threshold limit, the call is not queued and busy tone is returned to the transferring party.

Feature packaging

This feature is packaged under French Type Approval (FRTA) package 197 and Group Hunt/DN Access to SCL (PLDN) package 120.

Feature implementation

LD 57 – For the Group Hunt Queuing Limitation Enhancement, responses to the following prompts are required.

Prompt	Response	Description
REQ	CHG NEW	Modify or create data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Pilot DN.
PLDN	xxxx <CR>	Enter Pilot DN to be modified or created. Enter a carriage return to proceed to next prompt
USE	 GPHT	Enter use for Pilot DN. Group Hunting.
LSNO	xxxx	Enter group hunt list number. Group hunt list must exist in LD 18.
HTYP	(LIN) RRB	Enter either (Linear) or Round Robin as the type of hunting to be used for the group hunt list.
CFWI	(NO) YES	Call Forward All Calls Idle: enter NO if Group Hunting is to skip idle stations with Call Forward All Calls active, or enter YES if Group Hunting is to terminate on idle stations with Call Forward All Calls active.
MQUE	ACTM	Maximum Queue (maximum number of calls allowed to queue against the Pilot DN.): enter ACTM (Active Members) to limit the number of calls allowed to queue to be less than or equal to the number of active members of the group hunt list.

Feature operation

A group hunt list member is active if any call to the PLDN can terminate on the member set when it is idled. Conversely, a group hunt list member is not active if Group Hunt Termination Denied (GHTD) Flexible Feature Code (FFC) is dialed, and, or, Call Forward All Calls is active for the member and Call Forward Ignore (CFWI) in LD 57 is NO for the PLDN.

When the response to the MQUE prompt is ACTM and a call is routed to or dials a PLDN and it cannot terminate on an active member station, the call is linked to the PLDN queue (if the number of calls waiting in the PLDN queue is lower than the threshold limit). If the number of calls waiting in the PLDN queue reaches the threshold limit, calls are no longer linked to the PLDN queue. If the call is an internal call or attendant-extended call, busy tone is given to the originating party. If the originating call is a Direct Inward Dialing (DID) or Central Office (CO) trunk, it is routed to the attendant as a Call Forward Busy call. The Attendant Console display shows the PLDN (the attendant cannot Break-in or Busy Verify to a PDLN).

When a call is queued against a PLDN, Camp-on tone is given to the first member of the group hunt list having Warning Tone Allowed (WTA) Class of Service. If none of the members has WTA Class of Service, the Camp-on tone is not provided.

Handset Volume Reset

Contents

This section contains information on the following topics:

Feature description 477
Operating parameters 478
Feature interactions 478
Feature packaging 478
Feature implementation 478
Feature operation 478

Feature description

This feature is supported by the A44 chip in Meridian digital sets and causes a telephone's handset volume to be reset to a specified volume every time that the telephone user hangs up or uses handsfree. If the user wishes to adjust the volume, the user must manually do so for each call.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

LD 17 – Define the Handset Volume Reset setting.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN ATRN	Configuration Record. Aries Transmission.
...		
ATRN	YES	Aries (Meridian Modular) transmission parameters. Only prompted if response to TYPE is CFN.
...		
- VOLR	(NO) YES	Volume Reset.

Feature operation

When a transmission download occurs, following a SYSLOAD or when the set line cord is plugged in, the option setting defined in LD 17 is included in the message. The message is interpreted by the set firmware and the appropriate setting is applied. A system initialization will not download this message.

Handsfree Transmission Parameter Download

Contents

This section contains information on the following topics:

Feature description 479
Operating parameters 479
Feature interactions 480
Feature packaging 480
Feature implementation 480
Feature operation 483

Feature description

This feature provides parameters to support the handsfree transmission parameter download on Meridian 1 proprietary telephones. These parameters are downloaded to each telephone upon system reload or set power-up, after the handset parameters.

Two prompts are defined in LD 17 allowing control of handsfree transmit and receive loudness ratings.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Modify the system hardware and software parameters.

2 LD 17 – Create or modify the digital telephone data blocks.

3 LD 22 – Print Handsfree transmission parameter download settings.

LD 11 – Modify the system hardware and software parameters.

Prompt	Response	Description
...		
CLS	(HFD) HFA	Digital Telephone Handsfree (denied) allowed. Note: Not allowed on M2006, M2008, M2016S, or M2216 sets. M2016 must be defined as a 2616 with HFD Class of Service allowed for M2018 and M2616 sets. HFA is the default for M2317 and M3000 sets.

LD 17 – Create or modify the digital telephone data blocks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN ATRN	Configuration Record. Aries Transmission.
ATRN	YES	Meridian Modular telephone transmission parameters. Only prompted if response to TYPE is CFN.
- HRLR	(0)-8 32-40	Handsfree receive objective loudness rating.
- HTLR	(0)-11 32-54	Handsfree transmit objective loudness rating.

LD 22 – Print Handsfree transmission parameter download settings.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	CFN ATRN	Configuration. Meridian Modular telephone transmission parameters. ***Note:*** When the Handsfree transmission download parameters are printed they are output as their dB value (that is, an input of 0 in response to the HRLR prompt is printed as +42.00, while a response of 0 to the HTLR prompt is printed as -44.00).

Input value	HRLR (dB)	HTLR (dB)
0	+42.00	-44.00
1	+42.85	-43.50
2	+43.70	-43.50
3	+44.55	-43.00
4	+45.40	-42.50

Input value	HRLR (dB)	HTLR (dB)
21	N.A.	N.A.
22	N.A.	N.A.
23	N.A.	N.A.
24	N.A.	N.A.
25	N.A.	N.A.

Input value	HRLR (dB)	HTLR (dB)
5	+46.25	-42.00
6	+47.10	-42.00
7	+47.95	-41.50
8	+48.80	-41.00
9	N.A.	-40.50
10	N.A.	-40.50
11	N.A.	-40.00
12	N.A.	N.A.
13	N.A.	N.A.
14	N.A.	N.A.
15	N.A.	N.A.
16	N.A.	N.A.
17	N.A.	N.A.
18	N.A.	N.A.
19	N.A.	N.A.
20	N.A.	N.A.
42	N.A.	-48.00
43	N.A.	-48.00
44	N.A.	-48.50
45	N.A.	-49.00
46	N.A.	-49.50

Input value	HRLR (dB)	HTLR (dB)
26	N.A.	N.A.
27	N.A.	N.A.
28	N.A.	N.A.
29	N.A.	N.A.
30	N.A.	N.A.
31	N.A.	N.A.
32	+42.00	-44.00
33	+41.15	-44.50
34	+40.30	-45.00
35	+39.45	-45.00
36	+38.60	-45.50
37	+37.75	-46.00
38	+36.90	-46.50
39	+36.05	-46.50
40	+35.20	-47.00
41	N.A.	-47.50
53	N.A.	-52.00
54	N.A.	-52.50
55	N.A.	N.A.
56	N.A.	N.A.
57	N.A.	N.A.

Input value	HRLR (dB)	HTLR (dB)
47	N.A.	-49.50
48	N.A.	-50.00
49	N.A.	-50.50
50	N.A.	-51.00
51	N.A.	-51.00
52	N.A.	-51.50

Input value	HRLR (dB)	HTLR (dB)
58	N.A.	N.A.
59	N.A.	N.A.
60	N.A.	N.A.
61	N.A.	N.A.
62	N.A.	N.A.
63	N.A.	N.A.

Note: All values are Objective Loudness Ratings (OLR) measured without inserted loss or gain for trunk card interfaces and computed per IEEE methods. Receive ratings are at maximum volume. Transmit ratings are measured in an anechoic environment with less than 25 dBa room noise.

Feature operation

Whenever a download occurs, following SYSLOAD or when the telephone line cord is plugged in, the Relative Loudness Rating settings defined in LD 17 are included in the message. The message is interpreted by the telephone firmware and the appropriate settings are applied.

Held Call Clearing

Contents

This section contains information on the following topics:

Feature description 485
Operating parameters 486
Feature interactions 486
Feature packaging 487
Feature implementation 488
Feature operation 488

Feature description

The Held Call Clearing feature allows both the active call and the held call to be released when the user of a Meridian 1 proprietary telephone replaces the handset. Pressing the Release key only releases the active call.

For Single Appearance DNs, an on-hook action from a station clears the active call and all held calls belonging to that station. Pressing the Release key clears only the active call on the station. Activated feature keys, not involving an active or held call on the set, are not affected by the on-hook or Release key action. If an on-hook action occurs while a feature key is being activated, the system follows the Release key functionality. In most cases, this causes the feature key to be idled.

Where several DNs appear on the same set, an on-hook or Release key action does not affect any unanswered incoming calls which are unanswered call waiting calls or are in a ringing state, whether or not the ringing tone is audible. Answered call waiting calls (those which are active or being held) are cleared by an on-hook action. A Release key action clears only active call-waiting calls.

For Multiple Appearance DNs, an on-hook action from a station having one appearance of a Multiple Appearance DN clears only the current active call and the held calls belonging to that station. Pressing the Release key clears only the active call on the station. Calls active or held on another appearance of the same DN, on a different set, are not affected.

For Data DNs, an on-hook or Release key action clears active data calls on a Data DN. A data call is considered active on a set when the “Data Shift” LED is lit. A call on a Data DN which is not the set's active call is not affected by an on-hook or Release key action. For data terminals, only active data calls are released by an on-hook or Release key action.

Operating parameters

The Held Call Clearing feature cannot be used on analog (500/2500 type) telephones.

Feature interactions

Call Park

A call put on hold during a Call Park is not cleared by an on-hook action on that set.

Call Transfer

Active Call Transfer calls are cleared by either an on-hook or Release key action. Held Call Transfer calls are cleared only by an on-hook action, and not by a Release key action.

Called Party Control on Internal Calls

With Called Party Control on Internal Call enabled, a call on hold is not cleared when the calling party releases. This occurs whether or not the Held Call Clearing feature has been activated.

Conference

Active Conference calls are cleared by an on-hook or Release key action. Conference calls being held are cleared by an on-hook action only, and not by a Release key action. In either case, all other parties on the conference remain connected.

Handsfree

For a set equipped with a Handsfree add-on unit, the on-hook action is suppressed if the Handsfree key is pressed simultaneous to the on-hook. In this case, all active and held calls on the set are not affected by the on-hook action. For a Meridian M1000 or digital telephone, an on-hook action does not affect an active call on the set. In all cases, a Release key action clears an active call, whether in handsfree mode or not.

Misoperation on Call Transfer

An on-hook action clears a call that is put on hold during Call Transfer. This action may lead to a misoperation if the user of the set from which the call is being transferred goes on-hook before a valid DN is dialed. In this case, the misoperation is handled in the same manner as for a 500-type set.

On Hold on Loudspeaker

Going on-hook when Held Call Clearing is activated will clear the loudspeaker as for a normal held call. Therefore, it is recommended not to use this feature with the On Hold on Loudspeaker feature.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

LD 15 – Activate Held Call Clearing in response to the HCC prompt to implement this feature.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options
...		
- HCC	(NO) YES XFER	Held Call Clearing is to be activated, (deactivated) or set to transfer the held call.

Feature operation

Place the handset of your Meridian 1 proprietary telephone on-hook to release both the active and held call.

Note: Pressing the **Rls** key only releases the active call.

History File

The History File provides the capability to allocate an area of protected data to store system messages until a printout is requested by a technician. The size of the History File is defined on a system basis and can be up to 65 534 characters. Since one word of protected data stores two History File characters, the size of the History File is up to 32 767 words of protected data.

For a complete description of the History File, please refer to *System Management* (553-3001-300).

Hong Kong Digital Trunk Interface

Contents

This section contains information on the following topics:

Feature description 491
Operating parameters 492
Feature interactions 492
Feature packaging 492
Feature implementation 492
Feature operation 493

Feature description

This feature modifies the 1.5 Mbps Digital Trunk Interface (DTI) in order to allow the system to interface with the Hong Kong Telephone Company (HKT). The design modification alters the Dual-tone Multifrequency (DTMF) signaling protocol to conform with the HKT requirements. This DTMF design modification involves altering the AB bit protocol used in the DID/TIE convention, which is the convention used for the system to HKT connectivity. The AB bit values for the normal DID/TIE convention are reversed for the HKT interface. For example, if the normal convention for a DID/TIE going off-hook requires that AB bit values 0 and 0 be sent to the far end, the convention for HKT is that AB bit values 1 and 1 be sent.

This feature also meets the requirement of requiring the system, after a trunk seizure, to wait 600 milliseconds before accepting the dialed digits from the far end. This 600 milliseconds dialing delay is provided by the Dial Delay Timer, whose maximum configurable delay has been extended to 1,023 milliseconds. The timer is set on a per-route basis.

Operating parameters

Hong Kong Digital Trunk Interface modification applies only to 1.5 Mbits DTI trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under 1.5 Mbps Digital Trunk Interface (DTI) package 75.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Respond to the CLS prompt by entering HKA to allow the Hong Kong feature modification.

2 LD 16 – Enter a dialing pause for the Dial Delay Timer at the Time prompt:

LD 14 – Respond to the CLS prompt by entering HKA to allow the Hong Kong feature modification.

Prompt	Response	Description
...		
CLS	(HKD) HKA	Hong Kong DTI (denied) allowed. May only be used with DTI TNs with DTN CLS on DID or TIE routes.

LD 16 – Enter a dialing pause for the Dial Delay Timer at the Time prompt:

Prompt	Response	Description
...		
- TIMR	DDL 0-(70)-511	Dial Delay Timer. A value of 0 disables the timer.

Feature operation

No specific operating procedures are required to use this feature.

Hot Line, Enhanced

Contents

This section contains information on the following topics:

Feature description 495
Operating parameters 496
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Feature implementation 501
Feature operation 505

Feature description

Enhanced Hot Line (EHOT) provides Hot Line services to telephones with programmable keys. This feature is designed for, and is compatible with, analog (500/2500 type) telephones and Meridian 1 proprietary telephones. All capabilities from Flexible Hot Line (HOT) are provided to any key/lamp pair for one- and two-way Hot Lines on a per station basis. When the handset is lifted, or when a preprogrammed key is activated, the system speed calls a preprogrammed DN. Hot Lines access a set of terminal numbers programmed by direct entry using LD 11, or by list entry such as by System Speed Call (SSC) using LD 18. There is no difference in operation for the Hot Line user.

Once a Hot Line call enters the ringing state, it is the same as any normal call.

Enhanced Hot Line (EHOT) allows a distinction between analog (500/2500 type) telephone Hot Lines and manual Hot Lines without dial capabilities. For example, telephones with EHOT enabled and dial facilities support Dial Access features such as Call Transfer or Conference calling.

A Hot Line key can be defined with a Directory Number (DN) of its own, allowing other calls to terminate on that HOT key. The DN must be defined before it can be specified as the DN for a HOT key. For Meridian 1 proprietary telephones, the HOT key must be assigned to a DN during Service Change to create a two-way Hot Line. Analog (500/2500 type) telephones are always two-way Hot Lines, as they always have a DN assigned.

Operating parameters

Incoming calls to Hot Line telephones or keys can be restricted to calls originating from other Hot Line telephones or keys, Voice Call keys, and Group Call keys. This restriction is turned on or off on a per customer basis.

Telephones without a keypad or rotary dial cannot be assigned the Enhanced Hot Line Allowed (EHTA) Class of Service.

A maximum of 31 digits can be stored against a Hot Line telephone or key.

Only one Hot Line list is allowed per customer.

HOT cannot access a list created by the list-entry method for Enhanced Hot Line (EHOT).

A specific Hot Line key on a Meridian 1 proprietary telephone can have access to only one entry in the Hot Line list, but more than one telephone can have access to the same entry.

Analog (500/2500 type) telephones with Manual Line (MNL) Class of Service cannot be defined as Enhanced Hot Line Allowed (EHTA); Enhanced Hot Line Denied (EHTD) is the default. Users of these telephones must continue to use the HOT feature.

If a key is assigned as an EHOT Directory Number (DN), all appearances of that DN must also be EHOT keys.

Feature interactions

Attendant Administration

Use of an Attendant Console to change the database for EHOT is not supported.

Autodial

Flexible Hot Line and/or Enhanced Hot Line are mutually exclusive with the Autodial feature.

Automatic Answerback

The Automatic Answerback feature is fully compatible with a two-way Hot Line key assigned as the Prime DN.

Automatic Call Distribution

A Hot Line DN key can be assigned to an Automatic Call Distribution telephone.

Automatic Line Selection

Since the Hot Line key acts as a Single Call Ring (SCR) key, incoming ringing line preference can be applied. Outgoing line preference automatically selects a line other than the current Hot Line, so that a Hot Line call is not accidentally activated.

Automatic Redial

An Automatic Redial (ARDL) call can be activated from an Enhanced Hot Line key. However, the call is only redialed when the calling party's HOT key is free.

Call Forward Busy
Call Forward No Answer
Hunting

Any Hot Line telephone can be assigned Call Forward Busy, Call Forward No Answer and Hunting (excluding Short Hunt) Class of Service, but it applies only to the two-way Hot Line capability.

Call Park

Analog (500/2500 type) Hot Line telephones with EHTA and XFA Class of Service are allowed to park calls using the established Call Park procedures. Once a call is parked on an analog (500/2500 type) Hot Line telephone and the telephone is placed on hook, it cannot be unparked. Parked calls will recall to the parking telephone after the Call Park timeout. Two-way Meridian 1 proprietary telephone Hot Line stations that are equipped with a Call Park key/lamp pair are allowed to park calls in the normal fashion. As with analog (500/2500 type) telephones, a call parked from a Hot Line key cannot be picked up using the same key.

Call Pickup

Telephones with two-way Hot Line keys, and analog (500/2500 type) Hot Line telephones, can be assigned to pickup groups. Incoming Hot Line calls may be picked up by group members. To prevent someone from picking up a Hot Line call, do not put the Hot Line user into a Call Pickup group.

China – Flexible Feature Codes - Busy Number Redial

Busy Number Redial cannot be used on Enhanced Hot Line sets.

Controlled Class of Service

When a Hot Line DN is on a telephone that has Controlled Class of Service activated, Hot Line calls ignore the imposed Class of Service if the System Speed Call (SSC) package is present and the Hot Line list is given an adequate Network Class of Service (NCOS) for the override.

Dial Intercom

The analog (500/2500 type) Hot Line telephones cannot be members of Dial Intercom Groups (DIGs).

Digit Display

A Display key on a telephone with a Hot Line appearance will display the Hot Line target DN data stored for that key.

Display of Calling Party Denied

Display information on sets in a Hot Line call is based on the individual Class of Service of each set.

Enhanced Flexible Feature Codes - Busy Number Redial

The Busy Number Redial feature cannot be used on Enhanced Hotline sets.

Group Call

Hot Lines can be members of a Group Call. They cannot, however, have a Group Call key.

HOT

EHOT and HOT are mutually exclusive. A telephone cannot have both MNL and EHTA Classes of Service.

Internal Call Detail Recording

Hot Line stations can be assigned the appropriate Class of Service that allows Call Detail Recording records to be printed for calls originating on that telephone.

Make Set Busy

Make Set Busy is overridden by the Hot Line feature. If a Meridian 1 proprietary telephone is in Make Set Busy mode, incoming Hot Line calls still terminate (ring) on the telephone.

> ***Note:*** The Conference-Hot Line key overrides Make Set Busy only when the terminating key is HOT.

Override

A Hot Line call can be entered using the Override feature.

Permanent Hold

Analog (500/2500 type) telephones with EHTA cannot have Permanent Hold.

Prime Directory Number

If the Hot Line key is assigned to key 0 on a Meridian 1 proprietary telephone, it acts as the prime DN. When the user goes off-hook without selecting a DN key, the Hot Line is activated and the call is placed without further user action.

Private Line

A Hot Line key cannot be a Private Line, as this would defeat the benefits of Private Line service.

Room Status

The Room Status feature is incompatible with any telephone for which going off-hook activates Hot Line.

Speed Call, System

When the System Speed Call (SSC) package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

User Selectable Call Redirection

An analog (500/2500 type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Voice Call

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line.

When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Feature packaging

Enhanced Hot Line (HOT) package 70 requires:

- Network Class of Service (NCOS) package 32, and
- System Speed Call (SSC) package 34.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Assign the number of Speed Call lists, including Hot Line lists.

2 LD 15 – Add or change Enhanced Hot Line for a customer.

3 LD 18 – Use this prompt sequence to determine if there are enough memory and disk records for new Speed Call lists. Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.

4 LD 18 – Add or change a Hot Line Speed Call list.

5 LD 10 – Add Enhanced Hot Line for analog (500/2500 type) telephones.

6 LD 11 – Allow or deny Enhanced Hot Line for Meridian 1 proprietary telephones.

LD 17 – Assign the number of Speed Call lists, including Hot Line lists.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System Parameters.
MSCL	0-8191	Maximum number of Speed Call lists.

LD 15 – Add or change Enhanced Hot Line for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
OPT	(HTU) HTR	Hot Line (unrestricted) or restricted. This program determines whether the call is going to a Hot Line DN or to any available DN. HTR restricts Hot Line calls to Hot Line DNs, but HTU does not.

LD 18 – Use this prompt sequence to determine if there are enough memory and disk records for new Speed Call lists. Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8191	Number of lists to be added.
DNSZ	4-16-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of DN entries in Speed Call list.

LD 18 – Add or change a Hot Line Speed Call list.

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove a Speed Call list.
TYPE	HTL	Hot Line List.
CUST	xx	Customer number, as defined in LD 15
LNSO	0-8190	Hot Line List number (only one Hot Line List per customer).
NCOS	0-99	NCOS to be assigned to calls accessing the list.
DNSZ	xx	Maximum number of digits in a list entry (4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
STOR	xxx yy...y	xxx = list entry number (0-9, 0-99, or 0-999). yy...y = digits to be stored against the entry (must be equal to or less than DNSZ).
- WRT	(YES) NO	Data (is) is not correct and list (can) cannot be updated. The WRT prompt follows SIZE and STOR prompts asking for confirmation of the data just entered. If data is correct, enter YES or <CR>. A response of NO to WRT after SIZE returns the REQ prompt. A response of NO to WRT after STOR causes the data just entered to be ignored and a restart message (SCH3213) to be generated. A response of **** aborts the program. The last STOR value is lost but all other values for which WRT was YES are saved. The following information is output with the WRT prompt: ADDS: MEM: xxxxx DISK: yy.y (xxxxx is the amount of protected memory; yy.y is the number of disk records required for the new speed call list. Check the MEM AVAIL and DISK REC AVAIL values output before the REQ prompt).

LD 10 – Add Enhanced Hot Line for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	DTN DIP EHTA (LDN) LNA (XFD) XFA (CWD) CWA (XRD) XRA	Digitone or dial pulse service (manual service is not allowed). Enhanced Hot Line allowed. Last Number Redial (denied) allowed – optional. Call Transfer (denied) allowed – optional. Call Waiting (denied) allowed – optional. Ring Again (denied) allowed – optional.
FTR	HOT D nn x...x HOT L 0-999	Direct Hot Line DN. nn = number of digits (1-31) for target DN x...x. Hot Line List entry number defined in LD 18.

LD 11 – Allow or deny Enhanced Hot Line for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	nn HOT D cc x...x nn HOT L aaa nn HOT D cc x...x xxxx nn HOT L aaa xxx...x nn CH D cc x...x nn CH L aaa	One-way Hot Line key. One-way Hot Line List key. Two-way Hot Line key. Two-way Hot Line List key. Combined No Hold Conference and Direct Hot Line feature Combined No Hold Conference and Hot Line List feature nn = key number. cc = number of digits for target DN (1-31). x...x = target DN (up to 31 digits). aaa = Hot Line List entry defined in LD 18. xxx...x = DN for Hot Line key.

Feature operation

To make an EHOT call on an analog (500/2500 type) telephone:

- Lift the handset. The Hot Line number is automatically dialed.
- To transfer or conference an EHOT call on analog (500/2500 type) telephones:
- Flash the switchhook (or press **Link**) and dial the third-party extension.
- To make an EHOT call on a Meridian 1 proprietary telephone:
- Press **Hotline**.
- To answer an incoming Hot Line call on a Meridian 1 proprietary telephone:
- Press the flashing **Hotline** key.

To end an Enhanced Hot Line call:

- Hang up or press **Rls**.

Hot Line, Flexible

Contents

This section contains information on the following topics:

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Feature packaging 511
Feature implementation 511
Feature operation 512

Feature description

Flexible Hot Line (HOT) allows designated analog (500/2500 type) telephones to place calls to a predetermined destination simply by lifting the handset. The destination may be internal or external to the system, and the call does not require attendant intervention.

Flexible Hot Line (HOT) is provided to designated analog (500/2500 type) telephones on a Class of Service basis. A telephone is assigned the Hot Line feature through Service Change and a Manual Line (MNL) Class of Service. Address digits must be stored for the predetermined destination. If no digits are defined, the call will route to the Attendant Console.

When the user lifts the handset, no dial tone is returned. The system translates the stored digits and performs one of two operations:

- It rings an internal Directory Number (DN), then returns ringback tone.
- It translates to an external Trunk Access Code (TRC) and DN, then returns external call-progress tones or announcements.

Flashing the switchhook at any time during call setup or during the call will be ignored.

If the caller is a Hot Line, the prime Directory Number of the calling telephone is displayed on the terminating telephone, if equipped with a display.

Operating parameters

Flexible Hot Line applies to analog (500/2500 type) telephones only.

Feature interactions

Autodial

Flexible Hot Line and/or Enhanced Hot Line are mutually exclusive with the Autodial feature.

Calling Party Privacy

A Hot Line call will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Hot Line DN. The CPP will count against the maximum number of digits (currently 31) allowed for the Hot Line DN.

China – Flexible Feature Codes - Busy Number Redial
Enhanced Flexible Feature Codes - Busy Number Redial

Busy Number Redial cannot be used on Flexible Hot Line sets.

Conference

A Flexible Hot Line (non-enhanced) telephone cannot place conference calls, but an Enhanced Hot Line telephone can activate the conference feature. If the Hot Line restriction option is set, the conference call can terminate only to other Hot Line telephones. If the restriction option is not set, the conference call can terminate to any type of telephone

Enhanced Hotline

Flexible Hotline and Enhanced Hotline are mutually exclusive; a telephone cannot have both Manual Line (MNL) and Enhanced Hot Line Allowed (EHTA) Classes of Service.

EuroISDN Continuation

Flexible Hotline does not support EuroISDN Continuation.

Flexible Feature Code Boss Secretarial Filtering

Flexible Feature Code Boss Secretarial Filtering takes precedence over Private Line and Hot Line.

Hunting

Calls will hunt before being routed to the attendant. Any Hot Line telephone can be assigned Hunting (excluding Short Hunt) Class of Service, but it applies only to the two-way Hot Line capability.

ISDN QSIG/EuroISDN Call Completion

Call Completion cannot be used in conjunction with the Hot Line feature.

Make Set Busy

Make Set Busy is overridden by the Hot Line feature. If a Meridian 1 proprietary telephone is in Make Set Busy mode, incoming Hot Line calls still terminate (ring) on the telephone.

> ***Note:*** The Conference-Hot Line key overrides Make Set Busy only when the terminating key is HOT.

No Hold Conference

The Conference-Hot Line key supports only one-way Hot Line calls.

On Hold on Loudspeaker

It is possible to program Hot Line with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Override

A Hot Line call can be entered using the Override feature.

Phantom Terminal Numbers

Hot Line does not support Phantom Terminal Numbers.

Private Line Service

A Hot Line key cannot be a Private Line, as this would defeat the benefits of Private Line service.

Room Status

The Room Status feature is incompatible with any telephone for which going off-hook activates Hot Line.

Speed Call, System

When the System Speed Call package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

User Selectable Call Redirection

An analog (500/2500 type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Voice Call

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line. When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Feature packaging

The Flexible Hot Line feature is contained in Enhanced Hot Line (HOT) package 70. There are no feature package dependencies.

Feature implementation

LD 10 – Add or change Flexible Hot Line for analog (500/2500 type) telephones at the FTR prompt.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	MNL	Manual signaling – requires transfer denied (XFD) Class of Service.
FTR	HOT D 1-31 xxx...x yyy...y	Add Flexible Hot Line. 1-31 = maximum digits for Hot Line DNs. xxx...x = Flexible Hot Line DN. yyy...y = Phantom DN for a two-way Hot Line.

Feature operation

To make a Flexible Hot Line Call, follow these steps:

1 Lift the handset. The Hot Line number is automatically dialed.

2 To end the call, hang up.

Hunting

Contents

This section contains information on the following topics:

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Feature operation 526

Feature description

Hunting allows calls encountering a busy Directory Number (DN) to route automatically to another DN. Hunting continues along a predefined path, known as the hunt chain, until reaching an idle DN, the end of the hunt chain, or the maximum number of hunt steps. Hunting is specified on a DN basis. DNs in the hunt chain can be consecutive or nonconsecutive numbers.

The four types of hunt chains provided by the system are:

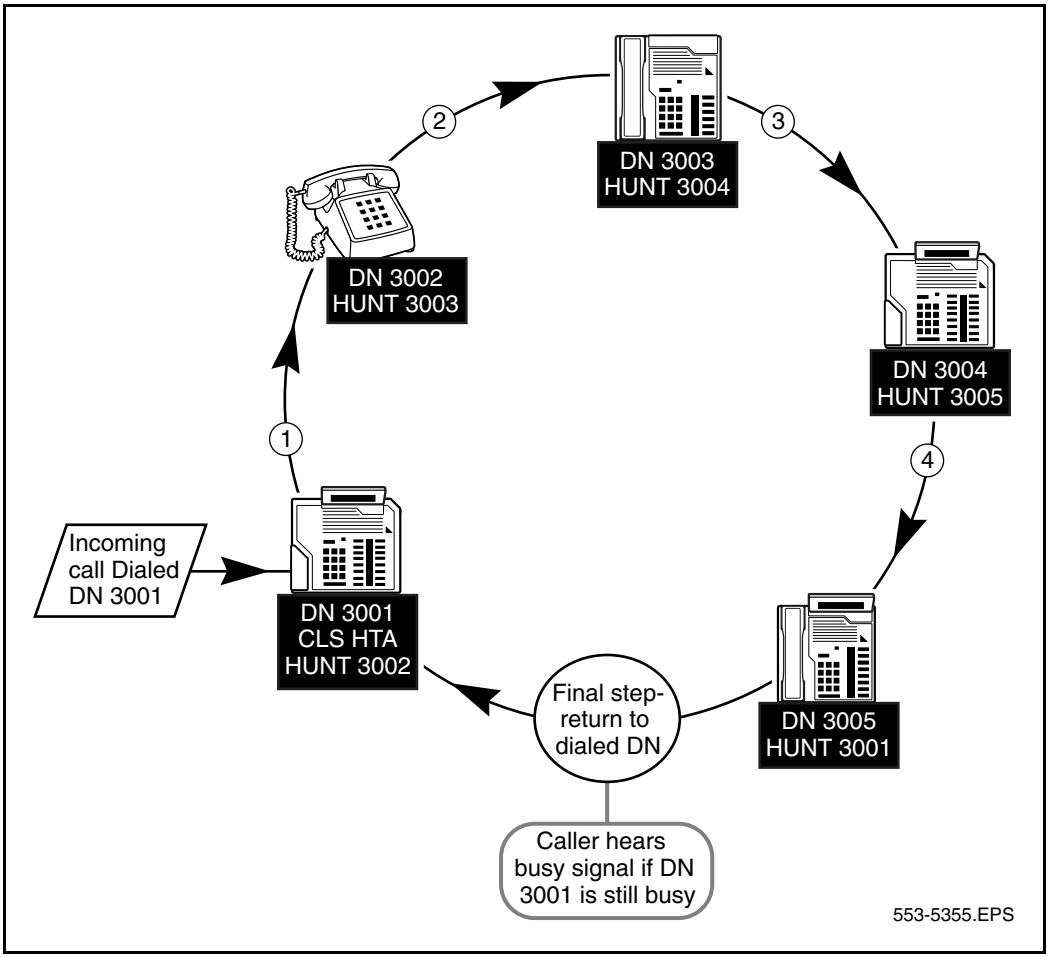
- Circular hunting
- Linear hunting
- Secretarial hunting
- Short hunting

The following pages describe and illustrate each of these ways to hunt.

Circular Hunting

Circular Hunting begins at the dialed DN and travels through every DN in the hunt group. The chain can begin at any point in the circle. The call goes around the circle until answered, or until returned to the initial DN. If all the DNs in the chain are busy, the caller hears busy tone. Figure 10 shows an example of circular hunting.

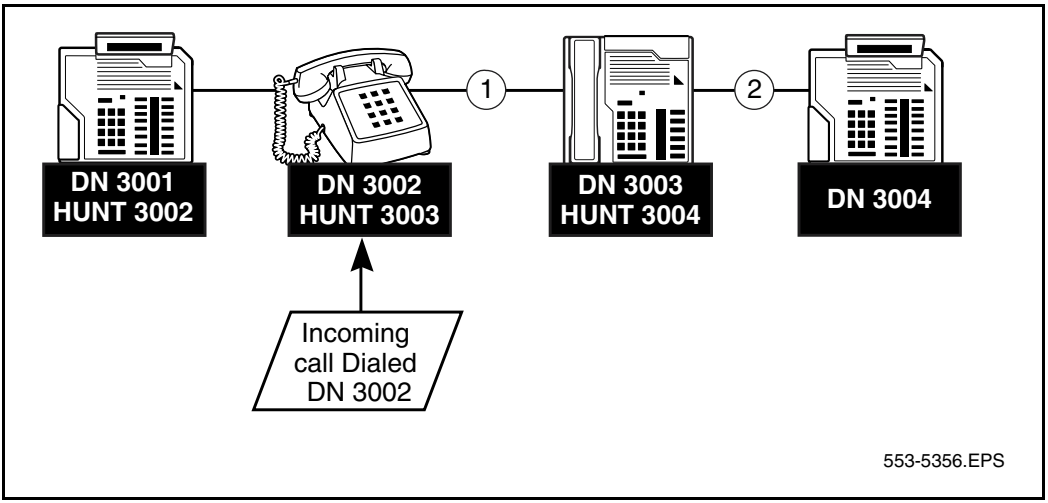
Figure 10
Example of Circular Hunting



Linear Hunting

Linear Hunting begins at the dialed DN. The call travels in one direction only when hunting along a linear chain. If a call comes into the second DN of a four-DN chain, it hunts to the third and fourth DNs only. If all the DNs are busy, the caller hears busy tone. Figure 11 shows an example of Linear Hunting.

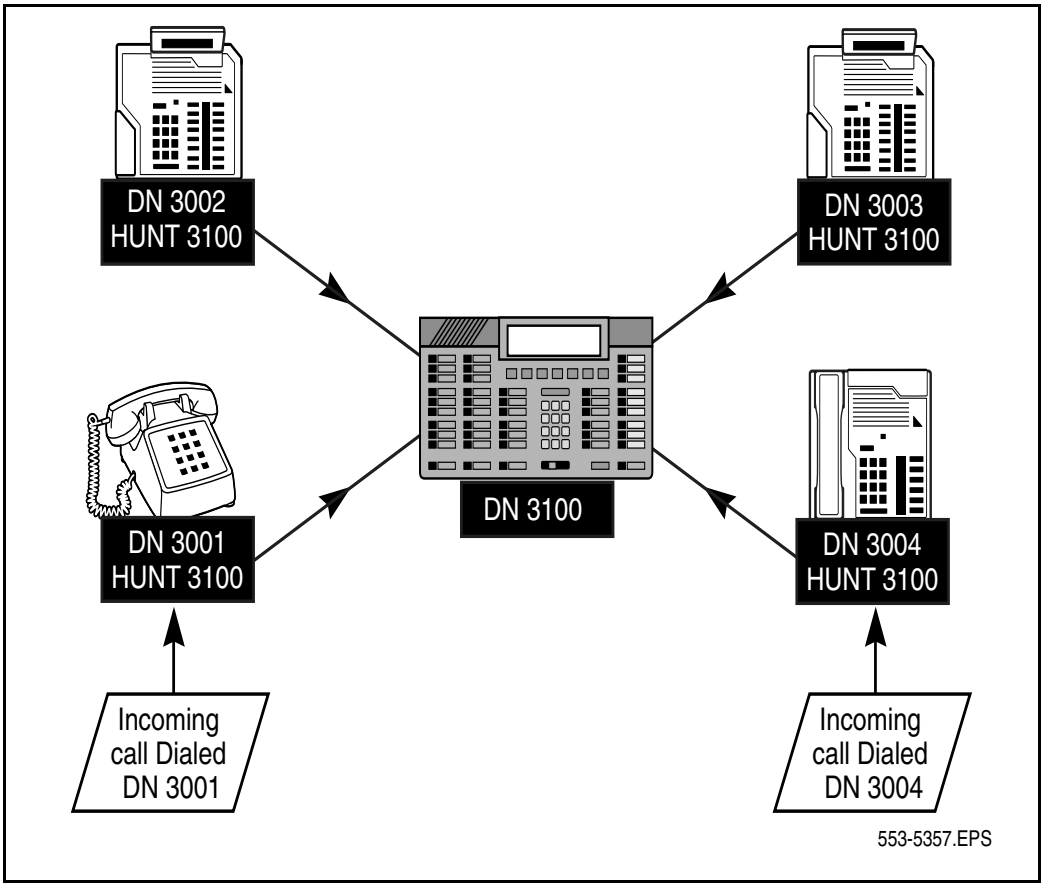
Figure 11
Example of Linear Hunting



Secretarial Hunting

Secretarial Hunting sends calls to a single Hunt DN, typically a secretary or Voice Mail. When a call comes in to a busy DN, it travels to the central location. Figure 12 shows an example of Secretarial Hunting.

Figure 12
Example of Secretarial Hunting



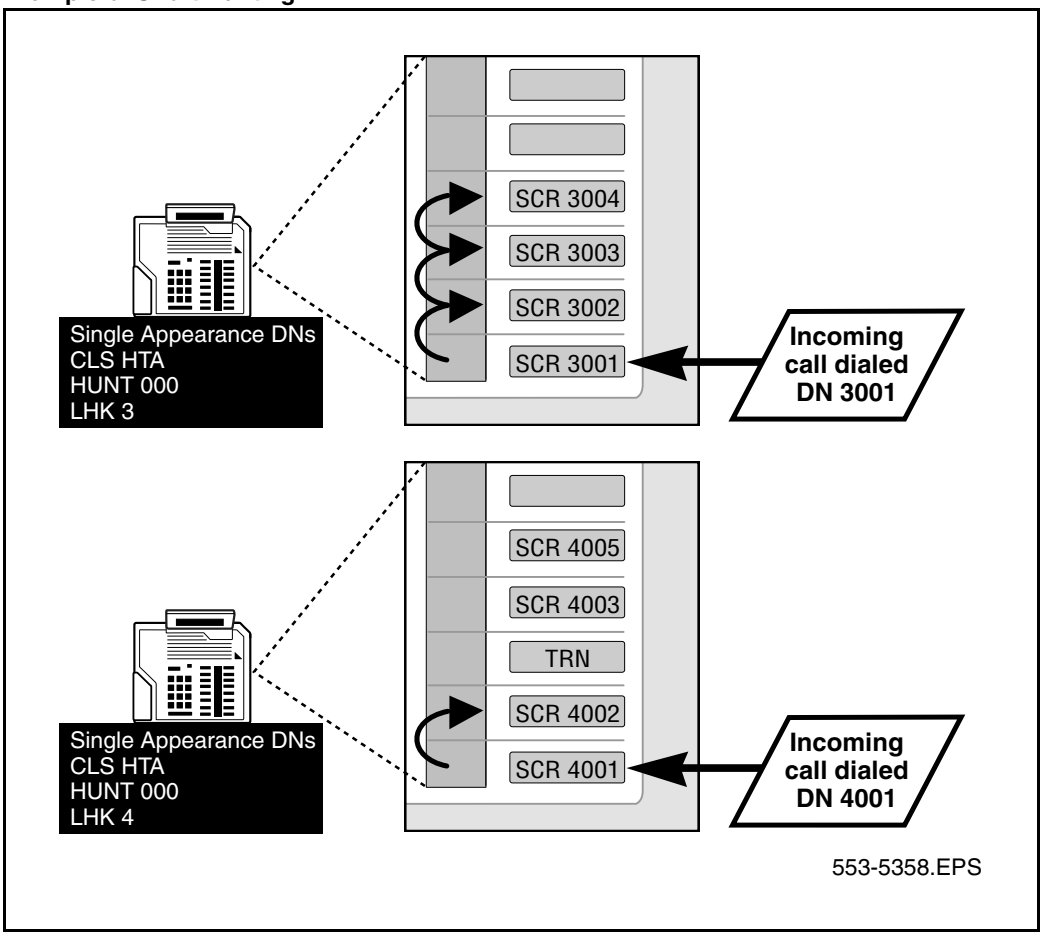
Short Hunting

Short Hunting takes place along the key strip of any Meridian 1 proprietary telephone. The hunt chain begins on a DN on the key strip. The call hunts up the keys until it reaches a feature key, an unassigned key, or the Last Hunt Key (LHK, defined in LD 11). If the call cannot reach an available DN, the caller hears busy tone. When a call hunts to a Multiple Appearance DN, all appearances with ringing are allowed.

For a TN with Hunting Control enabled, Short Hunt takes precedence over normal Hunting (Circular, Linear, or Secretarial). If the Hunting search selects a TN for a digital telephone, Short Hunt redirects the call before attempting to use the Hunt TN. The hunt chain might become Hunt DN A, Hunt DN B, Short Hunt Sequence C, Short Hunt Sequence D, or Hunt DN E.

Figure 13 shows an example of Short Hunting.

Figure 13
Example of Short Hunting



Operating parameters

There are no operating parameters associated with this feature

Feature interactions

Advice of Charge for EuroISDN

Calls charged with Advice of Charge that are either extended, transferred or redirected to another set using Hunting, are charged against the last station that answers the call and the controlling station releases.

Attendant Alternative Answering

Calls directed to a busy Attendant Alternative Answering (AAA) DN with Hunt defined are routed down the Hunt chain as defined for the AAA DN.

A Pilot DN for a hunting group can be defined as an AAA DN. Calls forwarded to a Pilot DN are directed to the next DN in the group.

Attendant Blocking of Directory Number

If Attendant Blocking of DN is attempted on a busy DN having the Hunting feature active, busy tone will be returned (overriding the Hunting feature).

Attendant Break-In

If the destination DN is in a Hunting chain with some idle DNs, the Break-In request goes to the first idle DN in the chain. To prevent this occurrence, the attendant can press the Break-In key prior to dialing the destination DN.

Attendant Busy Verify

Attendant Busy Verify does not affect Hunting.

Automatic Set Relocation

Calls will not hunt to a telephone that is being relocated.

Call Detail Recording on Redirected Incoming Calls

The Call Detail Recording on Redirected Incoming Calls feature does not affect how the Hunting feature operates; however, it does provide information about the answering party in the CDR ID field if incoming calls have been redirected by any one of these features.

Call Forward All Calls
Call Forward, Internal Calls

Call Forward All Calls and Internal Call Forward takes precedence over Hunting.

Call Forward Busy

Hunting takes precedence over Call Forward Busy for Direct Inward Dialing (DID) calls. When the station receiving a DID call has both Call Forward Busy and Hunting Allowed (HTA) Class of Service, the call is routed along the hunt chain. If all stations in the hunt chain are busy, the call is forwarded to the attendant.

Call Forward/Hunt Override Via Flexible Feature Code

A hunt can be overridden by the Call Forward/Hunt Override Via Flexible Feature Code feature, through the use of a Flexible Feature Code.

Call Forward No Answer, Second Level

A forwarded call may be modified by Hunting if the Call Forward No Answer DN is busy. This call is eligible for Second Level Call Forward No Answer if the SFA Class of Service is allowed and a Call Forward No Answer DN has been defined for the last rung DN.

If Group Hunting is active, Second Level CFNA is not applied.

Call Page Network Wide

Call Page Network Wide (PAGENET) does not block a station set from being programmed to Hunting to an external Paging trunk. At call termination time, calls that are forwarded to an external PAGENET uncontrolled trunk are not blocked. However, calls forwarded to an external PAGENET controlled trunk are given access denied intercept treatment at the Paging node.

Call Redirection by Time of Day

When Call Redirection by Time of Day (CRTOD) is enabled and an incoming call reaches a busy Directory Number, the time is checked against the Alternate Redirection Time Option range defined on the telephone.

Call Waiting
Station-to-Station Call Waiting

If a call comes into a busy DN, it begins the hunting route defined from the called DN. If there are idle DNs on the hunting route, the call becomes a Call Waiting call on the called DN.

Hunting takes precedence over Call Waiting. If all steps in the hunt chain are busy, Call Waiting is activated.

Call Waiting Redirection

If Call Forward and Hunt by Call Type (CFCT) is enabled with Call Forward No Answer and Call Waiting Redirection, “no answer” internal calls receiving Call Waiting treatment are routed for CFNA treatment to the Flexible CFNA DN (FDN) or Hunt DN, and “no answer” external calls are routed for CFNA treatment to the External Flexible CFNA DN (EFD) or External Hunt DN (EHT).

Calling Party Privacy

When an incoming trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number, provided that the tandem node also has Calling Party Privacy (CCP) provisioned.

If an incoming ISDN trunk call with the Privacy Indicator is forwarded, the Privacy Indicator will be tandemed to the far end to inhibit the display of the Calling Party Name or Number provided that the outgoing trunk route on the tandem node also has CCP provisioned.

If an incoming non-ISDN trunk call is forwarded to a trunk, the outgoing trunk call from the tandem node will carry the Privacy Indicator if the outgoing trunk route on the tandem node has the TCPP option set.

The CCP code can also be stored on the forwarding DN. If the CPP is requested on the forwarding DN, the Privacy Indicator will be outpulsed to the terminating node to inhibit the number of the forwarding set (that is, at the tandem node) from being displayed on the terminating set. In this scenario, the forwarding station must include the CPP in the forwarding DN (such as *67 + ACOD + the DN on the terminating node).

The above scenario also applies to Network Hunt.

Camp-On
Camp-On, Station

Hunting takes precedence over Camp-On and Station Camp-On.

Capacity Expansion

If more than 16 appearances of the same Directory Number (DN) are configured, each hunt step is counted as two, to avoid running out of time slots.

China – Toll Call Loss Plan

Toll pad switching is also provided after call hunting has been completed. When the toll call is diverted, the diverted party's pad level is switched back to its original value (unless it is an OPS station using dynamic switching). The Toll Loss Plan is applied again for the new call as if it is a direct call. For Call Transfer, it is provided after the transferring party completes the transfer and drops out. For Call Forward or Hunting, it is provided when the forwarding or hunting call is answered.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

If Executive Intrusion is attempted against an extension with a Hunt DN configured, an attempt will be made to reroute the call to the hunt DN provided the Hunt DN is on the same node. If the Hunt DN is busy, this rerouting process is repeated. If all DNs in the Hunt chain are busy, Executive Intrusion is attempted against the wanted extension originally dialed. Otherwise, the call will terminate as a simple call on the first idle extension in the Hunt chain.

Direct Inward Dialing Call Forward No Answer Timer

Hunting takes precedence over the Message Center feature.

Do Not Disturb

If activated, Hunting takes precedence over Do Not Disturb busy indication.

Flexible Feature Code Boss Secretarial Filtering

A boss set with filtering activated is passed over by Hunting; the next hunt sequence is to the secretary set.

Group Call
Dial Access to Group Calls

Hunting cannot be applied to a Group Call.

Group Hunt

Group Hunting has priority over Hunting. If the DN attempted for termination by Group Hunting has HTA COS, and if it is busy, Group Hunting continues with the next DN in the group instead of following the DN's hunting configuration.

Hot Line

Any Hot Line telephone can be assigned Hunting (excluding Short Hunt) Class of Service, but it applies only to the two-way Hot Line capability.

ICP Network Screen Activation
Flexible DN
Meridian Mail

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an Intercept Computer (ICP) position, a redirected message identification "50" is sent to the ICP computer, when the call is answered.

Idle Extension Notification

If the attendant dials a busy extension that has Hunting configured and where all the DNs in the hunt chain are busy, Idle Extension Notification may be requested towards the dialed extension.

ISDN QSIG Name Display

When an incoming QSIG call with name display presentation allowed is hunted locally, the calling party's name information is displayed on the destination set. With presentation restriction, the calling party's name information is not displayed.

Lockout, DID Second Degree Busy, and MFE Signaling Treatments
Multiple Appearance Directory Numbers

Hunting is controlled by the MADN Redirection Prime (MARP) Terminal Number (TN). If the MARP system option is disabled, Hunting proceeds as if MARP did not exist.

If all the telephones in the Multiple Appearance Directory Number (MADN) group are Meridian 1 proprietary telephones, ringing telephones are placed at the top of the DN list, and non-ringing telephones are placed at the bottom.

If a Multiple Appearance Directory Number appears in a group with several telephone types, the telephone type affects the position of the TN in the list. The analog (500/2500 type) telephones are listed at the top, and Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the top of the list. A service change to a Meridian 1 proprietary telephone moves it to the bottom of the list. Call redirection follows the TN order from top to bottom.

The MARP TN is always checked to determine if and how the call is to be redirected by Hunting, regardless of where the MARP TN resides in the TN list of the DN block. No searching of the TN list of the DN block is needed. Hunting will follow the hunt chain based on the originally dialed DN. The actual functioning and requirements for Hunting are not changed by the MARP feature. The basic change introduced by the MARP feature is to always have a designated TN, the MARP TN, as the TN supplying the call redirection parameters.

If the MARP TN does not have Hunting control enabled, no Hunting is attempted. Other features for redirecting calls to busy DNs may be attempted based on the MARP TN.

A Short Hunting sequence begins when the MARP TN of a busy DN can perform Short Hunting. When a Short Hunt begins, it completes on that telephone before going to the Hunt DN. The precedence of Short Hunting over normal Hunting is maintained. Once a Short Hunting sequence is started on a digital TN, all the DNs in the Short Hunt sequence on that TN are attempted before redirecting the call to the TN's Hunt DN. Thus, a Hunt Chain connects Short Hunting sequences through Hunt DNs only.

Multiple Appearance Directory Number Redirection Prime

The Multiple Appearance Directory Number Redirection Prime (MARP) TN always controls the call redirection for Hunting. Short Hunting takes precedence over Hunting and MARP. The MARP TN is referred to until Short Hunting is encountered. Short Hunting is in control until it expires. When short hunting expires, the MARP TN for the first DN in the Short Hunt sequence takes control.

Network Individual Do Not Disturb Recovery on Misoperation of Attendant Console

Hunting takes precedence over the Network Individual Do Not Disturb and the Misoperation feature.

On Hold on Loudspeaker

Hunting to a loudspeaker DN can be programmed, but will receive intercept treatment as for direct dial to the loudspeaker DN.

Recorded Announcement for Calls Diverted to External Trunks

Recorded Announcement for Calls Diverted to External Trunks (RANX) is activated if the call is forwarded to an outgoing external CO trunk with the RANX feature active.

Recovery on Misoperation of Attendant Console

Hunting takes precedence over the Misoperation feature.

Ring Again on No Answer

If Ring Again on No Answer has been applied to a station going through a Hunt sequence, Ring Again is applied to that station and not the ringing station.

Total Redirection Count

Hunt redirections is limited to the value defined in the Total Redirection Count limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

User Selectable Call Redirection

User Selectable Call Redirection permits a user to alter the HUNT DNs or EHT from a telephone.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Add or change Hunting for analog (500/2500 type) telephones.

2 LD 11 – Add or change Hunting for Meridian 1 proprietary telephones.

LD 10 – Add or change Hunting for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
HUNT	xxx...x	Hunt DN. xxx...x removes the DN from the hunt chain.
CLS	(HTD) HTA	(Deny) allow hunting.

LD 11 – Add or change Hunting for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
HUNT	xxx...x 000	Hunt DN. xxxx...x removes the DN from the hunting chain. Allow Short Hunting only.
LHK	xx	Last Hunt Key (LHK) number (default is 0). LHK 0 deactivates Short Hunt.
CLS	(HTD) HTA	(Deny) allow hunting.

Feature operation

No specific operating procedures are required to use this feature.

Hunting by Call Type

Contents

This section contains information on the following topics:

Feature description 527
Operating parameters 528
Feature interactions 528
Feature packaging 528
Feature implementation 529
Feature operation 529

Feature description

An additional Class of Service is provided for the system which will allow Direct Inward Dialing (DID) calls to hunt using the hunt chain when the dialed extension is busy, and the call's Classes of Service are Hunt by Call Type Deny (HTD) and Hunt by Call Type Allowed (HBTA).

The following rules apply to the call processing:

- If an extension is busy and its Class of Service is HTA, all types of calls to the extension will hunt using the hunt chain, regardless of HBTA/HBTD and FBA/FBD.

- If a busy extension's Class of Service includes HTD and HBTD, internal calls to the extension receive busy tone. Direct Inward Dialing (DID) calls to the extension which have Class of Service FBA are forwarded to the attendant. DID calls to the extension which have a Class of Service of FBD receive busy tone.
- If a busy extension's Class of Service include HTD and HBTA, internal calls to the extension receive busy tone. DID calls to the extension hunt using the hunt chain. If hunting fails, DID calls to the extension which have a Class of Service of FBA are forwarded to the attendant, and DID calls with a Class of Service of FBD receive busy tone.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Redirection by Time of Day

If Call Forward by Call Type (CFCT) is enabled with Call Forward No Answer (CFNA) and Call Redirection by Time of Day (CRTOD), unanswered internal calls receiving CFNA are routed to the Flexible CFNA DN, Hunt DN, Alternate Flexible CFNA DN or Alternate Hunt DNs. External calls are routed in the same manner.

If CFNA is enabled with Hunting by Call Type and Call Redirection by Time of Day (CRTOD), unanswered internal calls are redirected to the Hunt DN or Alternate Hunt DN during the alternative time. External calls are routed in the same manner. The alternate time is defined on the called DN’s data block.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Create or modify the analog (500/2500 type) telephone data blocks to allow/deny Hunt by Call Type.

2 LD 11 – Create or modify the Meridian 1 proprietary telephone data blocks to allow/deny Hunt by Call Type.

LD 10 – Create or modify the analog (500/2500 type) telephone data blocks to allow/deny Hunt by Call Type.

Prompt	Response	Description
...		
CLS	(HBTD) HBTA	Hunt by Call Type (denied) allowed.

LD 11 – Create or modify the Meridian 1 proprietary telephone data blocks to allow/deny Hunt by Call Type.

Prompt	Response	Description
CLS	(HBTD) HBTA	Hunt by Call Type (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Hunting, Data Port

Contents

This section contains information on the following topics:

Feature description 531
Operating parameters 532
Feature interactions 533
Feature packaging 533
Feature implementation 533
Feature operation 536

Feature description

Data Port Hunting improves the Hunting operation for data ports and modem pooling, and improves Ring Again operation for modem pooling.

Up to 255 data ports can be configured as trunks in data port trunk routes. In addition, the route can be programmed to step to another data port route if all members in the route are busy.

A data port serves as the interface between the system and a computer or other data communication device. A data port can be one of the following devices:

- Standalone Add-on Data Module (ADM) in auto-answer mode (no modem)
- Any modem that can recognize ringing and simulate off hook or on hook status
- Standalone ADM in auto-answer mode, connected to a modem

- Data Access Card (DAC), or
- Meridian Communications Adapter (MCA).

The following types of trunk routes are supported for data port hunting:

- ADM Trunk Routes: Add-on Data Module (ADM) data ports that interface through Data Line Cards
- Modem Trunk Routes: Modem data ports that interface through 500/2500 Line Cards
- RS-232 (R232): RS-232 data ports that interface through Data Access Cards (DACs)
- RS-422 (R422): RS-422 data ports that interface through Data Access Cards (DACs), and
- MCA: Meridian Communications Adapter (MCA) data ports that interface through Integrated Services Data Line Cards (ISDLCs) or Data Line Cards (DLCs).

Data ports act only as terminating parties. The user dials the access code of the trunk route to access the data ports.

Operating parameters

All data port trunks within a route must be of a single type. ADM and MDM data ports cannot be mixed in the same data port trunk route.

Only an attendant can extend incoming calls from stations or trunks (CO, FX, WATS, TIE, Direct Inward Dialing [DID], Common Controlled Switching Arrangement [CCSA]) to data port trunk routes. Calls cannot be extended, transferred, or conferenced from a station to a data port group.

In Night Service mode, any station can transfer incoming calls to data port routes.

Trunk access restrictions (TARG, TGAR) should be applied to data port trunk routes to prevent stations with co-located ADMs from directly accessing data ports with modems, and vice versa.

Class of Service restrictions do not apply to data port trunks.

Ring Again, Basic/Network Alternate Route Selection (BARS/NARS), and trunk access restrictions (TARG, TGAR) are the only features that may be applied on calls to data port routes.

Feature interactions

Conference

There are no feature interactions associated with this feature.

Ring Again

When a user activates Ring Again against the data port extension Access Code (ACOI), the system stores the request until a member in the data port route becomes idle. When an idle member is found, the calling party is notified and the member is reserved for eight seconds. If the calling party does not respond to the Ring Again notification within eight seconds, the reservation is dropped.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Add or change a data port trunk route.

2 LD 14 – Add or change a data port trunk.

LD 16 – Add or change a data port trunk route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.

TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	ADM MDM R232 R422 MMPM	Trunk route type
STEP	0-511	Alternate trunk route number
TARG	0-31	Trunk Access Restriction Groups (TARGs)
- TOV		Data port time out
	0-3	0 = No timeout 1 = 15 minutes 2 = 30 minutes 3 = 60 minutes
- PSEL	(DMDM) TLNK	Protocol selection. T-Link or DM-DM protocol. Prompt offered to MCU (TKTP = MMPM).
- OPE	(NO) YES	(Do not) change data port operating parameters. Prompt offered to MCU (TKTP = MMPM).
- - PSDS	(NO) YES	(Do not) allow PSDS protocol. Prompt offered to MCU (TKTP = MMPM).
- - TRAN	(ASYN) SYN	Port transmission type; if PSDS = YES, then TRAN must be SYN. Prompt offered to MCU (TKTP = MMPM).
- - PAR		Parity type, where:
	(SPAC) EVEN ODD MARK	SPAC = space parity EVEN = even parity ODD = odd parity MARK = mark parity
- - DTR	(OFF) ON	Forced DTR (if ON) or dynamic DTR (if OFF). Prompt offered to R232, and to MCU (TKTP = MMPM).
- - DUP	(FULL) HALF	Full duplex/half duplex. Prompt offered to MCU (TKTP = MMPM).

- - DCD	(ON) OFF	(ON) = dynamic CD. OFF = forced CD. Prompt offered to R232, and to MCU (TKTP = MMPM).
- - MOD	(NO) YES	Modem, (Network): when TRAN = SYN. Prompt offered to MCU (TKTP = MMPM).
- - INT	(OFF) ON	SL-1/100 Interworking. Prompt offered to MCU (TKTP = MMPM).
- - CLK	(OFF) ON	(OFF) = External Clock, ON = Internal, when TRAN = SYN. Prompt offered to MCU (TKTP = MMPM).
- - V25	(NO) YES	V.25 bis offered only when TRAN = SYN. Prompt offered to MCU (TKTP = MMPM).
- - HDLC	(NO) YES	High Level Data Link Control offered only when V25 = YES. Prompt offered to MCU (TKTP = MMPM).
- - DEM	(DCE) DTE	Data Equipment Mode. DCE or DTE mode. Prompt offered to R232.
- - PBDO	(OFF) ON	Port Busy upon DTR off. Presented when DCE, Dynamic DTR. Prompt offered to R232. ON = enabled (OFF) = disabled

LD 14 – Add or change a data port trunk.

Prompt	**Response**	**Description**
REQ	NEW CHG	New or change.
TYPE	ADM MDM R232 R422 MMPM	Trunk type
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

Feature operation

To access a Data Unit (DU), the user dials the Access Code (ACOD) of the route data block. If a DU is available, a connection is made. If a DU is unavailable, the user receives this message on the terminal screen: "ALL PORTS ARE BUSY. ACTIVATE RING AGAIN?" Select Ring Again and wait until a DU port becomes available.

When a user dials a data port, the request is placed in the Ring Again queue until a port becomes idle. When an idle port is located, the calling party is notified and the port is reserved for eight seconds.

Data Port Verification (DVS)

Any applicable telephone with Data Port Verification Allowed (ADV) Class of Service can place a call to a specific Add-on Data Module (ADM) in a route by going off-hook, receiving dial tone, and dialing:

SPRE + 70 + ACOD + mmm

where:

SPRE	=	special prefix
70	=	special access code for the Data Port Verification (DVS) feature
ACOD	=	Access Code for the ADM trunk group, and
mmm	=	three-digit number that is to be seized within the trunk group.

The selected ADM trunk is seized if it is in not busy, maintenance busy, or disabled state. Once the call is established, it is treated as a normal ADM trunk call. If the selected trunk is in busy, maintenance busy, or disabled state, the call originator receives an overflow tone. No tone is returned when keyboard dialing is used.

Hunting, Trunk

Contents

This section contains information on the following topics:

Feature description 537
Operating parameters 538
Feature interactions 539
Feature packaging 539
Feature implementation 539
Feature operation 539

Feature description

Trunk Hunting provides either Linear Hunting or Round Robin Trunk Hunting for outgoing trunks in a route.

When Linear Hunting is implemented, the system searches for an available trunk in descending order. A station originating an outgoing call is connected to the last available trunk (highest available trunk route member number) of the trunk route accessed. The last trunk route member is always the first choice for outgoing calls and the first trunk route member is always the last choice.

Round Robin Trunk Hunting

Outgoing calls are evenly distributed among the members of a trunk route. When a station originates an outgoing call, the system searches for an available trunk route member in descending order, starting with the next lower member number from the last trunk seized for an outgoing call on the trunk route. If a trunk with a lower member number is not available, the system searches for a trunk starting with the highest member number of the route.

Note for multiple group machines using Round Robin Trunk Hunting:

To minimize system resource usage, the system will attempt to hunt to an available trunk within the same group as the originating TN. For example, if a call is placed from a telephone whose TN is in group 1, the system will first attempt to locate an available trunk within group 1. If there are no available trunks in group 1, the system selects an available trunk from another group.

Each time hunting occurs, the round robin index value, which points to the next route member to be examined, is updated. Because the proximity of a trunk loop to the originating TN loop takes precedence over the order of the trunk route members, the system may be forced to hunt through many route members to locate an available trunk within a given group. This can cause the round robin index to change dramatically, yielding inconsistent trunk usage patterns.

If uniform trunk usage is a prime concern, configure route members with alternating groups. For example, if a given route contains trunk members from different groups, alternate the groups so that route member 1 is a trunk member from group 1, route member 2 is a trunk member from group 2, and so on. This configuration will produce more uniform trunk usage than would occur if trunks of the same group were bunched together within a route.

Operating parameters

The Public Exchange/Central Office (CO) governs incoming trunk hunting. The system has no control over the order of incoming trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 16 – Implement Linear or Round Robin Trunk Hunting for a trunk route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
SRCH	(LIN) RRB	Linear or Round Robin Hunting.

Feature operation

No specific operating procedures are required to use this feature.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

Contents

This section contains information on the following topics:

Feature description 541
Operating parameters 543
Feature interactions 544
Feature packaging 545
Feature implementation 546
Feature operation 547

Feature description

This feature provides the following enhancements to the Intercept Computer (ICP) feature:

- Network Screen Activation (NWSA) allows network-wide application of an ICP screen.
- Flexible DN Length (FXDN) allows the maximum length of DNs sent to the ICP to be seven digits (shorter DNs are still padded with zeros).
- Meridian Mail Interactions (MMIA) allow ICP and Meridian Mail to be configured for the same customer, by removing all interactions between them.

Network Screen Activation (NWSA)

Calls intended to terminate on one node but which are redirected to an ICP position using ICP Forward, Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt, are presented on the ICP terminal (ICT) at that position.

Direct calls from another node to an ICP position are presented on the ICT at that ICP position. Recalls to the ICP attendant are presented on the ICT at that ICP position attendant.

Calls which are made or extended by an ICP position attendant to another node, and which terminate at an ICP position attendant, follow Network Attendant Service (NAS) and Network ACD (NACD) treatment. If a call is rejected, it is presented on the ICT at the originating ICP position attendant. If a call terminates at an ICP position set, the call is established and presented on the ICT at the terminating ICP position set.

NWSA uses the definition of Call Forward by Call Type to perform call forwarding. All calls are forwarded to the Flexible Call Forward No Answer DN (FDN), if one has been defined; external calls are forwarded to the External DN (ECDN); private network calls are treated as internal calls and forwarded to the Internal DN (ICDN). In the case where a call is made or extended from a local or network ICP position attendant, the call is treated as an external call to avoid having it forwarded to the ICP answering machine.

The maximum number of digits for the FDN, ICDN and ECDN is 13.

Flexible DN Length (FXDN)

Since the standard maximum length of DNs in a system is seven digits, the maximum length of DNs sent to the ICP is seven digits. However, since some ICP computers can only handle a maximum DN length of four or five digits, flexibility has been provided by allowing an entry in LD 15 of between three to seven digits. The selected length must be fixed; DNs shorter than the selected length must be padded by a digit between zero to nine, also configured in LD 15.

Meridian Mail Interactions (MMIA)

Meridian Mail and ICP may be configured in LD 15 for the same customer number, by answering "YES" to both the IMS prompt and the ICP prompt. Meridian Mail and ICP can then be used by the same customer, independent of each other. A set may be configured to have its calls forwarded to Meridian Mail or the ICP, or a mixture of both (for example, all internal calls can be configured to be forwarded to Meridian Mail, by setting the ICDN or FDN to the Meridian Mail Message Center DN, and all external calls to be forwarded to the ICP intercept position by setting the ECDN to the ICP Message Center DN).

Operating parameters

For NWSA functionality:

- the ICP has to be connected to all nodes in the network
- the same requirements and limitations apply as for Network Call Redirection and Network Attendant Service, and
- ICP to network nodes connection, and network node to network node connections must be using Integrated Services Digital Network (ISDN) links.

For FXDN functionality:

- the DN sent to the ICP is the originally called station, or in the case of direct calls, is the calling station, and
- the length of DNs may differ from node to node; however, the node with the ICT must be configured for the maximum length within the network.

For MMIA functionality:

- ICP and Meridian Mail cannot use the same port; however, ICP and Meridian Mail may be configured on separate ports for the same customer number

- if a set has been configured to have call forwarding to both ICP and Meridian Mail, retrieving of messages by activating the Message Waiting key (MWK) can only be done for either ICP or Meridian Mail, and
- the Message Waiting lamp indication cannot support both ICP and Meridian Mail simultaneously (that is, if a set has been configured to have call forwarding to both ICP and Meridian Mail, and a call is waiting from both ICP and MM, the Message Waiting lamp goes dark after one of the messages has been retrieved from either ICP or MM).

Feature interactions

The same interactions apply as for the ICP feature, other than the ones between Meridian Mail and ICP. The interactions described below also apply.

Attendant Recall

When a call from another node is recalled to the ICP position attendant, it is presented on the ICP terminal.

Call Forward All Calls
Call Forward Busy
Call Forward No Answer
Hunting

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an Intercept Computer (ICP) position, a redirected message identification “50” is sent to the ICP computer, when the call is answered.

Electronic Switched Network)

The only Electronic Switched Network functionality which is supported is Coordinated Dialing Plan.

Network Call Redirection

For ICP-forwarded calls, the Network Call Redirection reason is Call Forward Unconditional.

Slow Answer Recall

When an attendant extends a call to a set with call forward active, the slow answer recall timer at the originating node will be reset for ICP forward.

Slow Answer Recall for External Transferred Calls

When an ICP position set transfers an external call across an ISDN network, the slow answer recall timer is set at the transferring node to prevent the terminating set to be rung indefinitely. When the slow answer recall timer times out, the transferred call is recalled to the attendant at the transferring node.

Feature packaging

The following packages are required for ICP Network Screen Activation, Flexible DN and Meridian Mail Interactions:

- Intercept Computer Interface (ICP) package 143
- Integrated Message Services (IMS) package 35
- Automatic Call Distribution Package A (ACDA) package 45
- Message Waiting Center (MWC) package 46
- Auxiliary Processor Link (APL) package 109; Flexible Feature Codes (FFC) package 139
- International Supplementary Features (SUPP) package 131

The following packages are also required for the NWSA enhancement:

- Integrated Services Digital Network (ISDN) package 145
- Advanced ISDN Network Services (NTWK) package 148
- 1.5 Mbit Primary Rate Access (PRA) package 146
- Network Attendant Service (NAS) package 159

The following package is also required for the FXDN enhancement:

- DN Expansion (DN) package 150

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – For NWSA, enter up to 13 digits for ICDN and ECDN.

2 LD 15 – For FXDN, set the DN length and any padding digits at the ICDL and ICPD prompts.

3 LD 15 – For MMIA, enter "YES" to both the Meridian Mail prompt (IMS) and the Intercept Computer prompt (ICP).

LD 15 – For NWSA, enter up to 13 digits for ICDN and ECDN.

Prompt	Response	Description
...		
TYPE:	ICP	Intercept computer update.
- ICP	YES	ICP is available.
...		
- ICMM	0-9	Message number.
- ICDN	0-13	Default internal DN.
- ECDN	0-13	Default external DN.

LD 15 – For FXDN, set the DN length and any padding digits at the ICDL and ICPD prompts.

Prompt	Response	Description
...		
TYPE:	ICP	Intercept computer update.
- ICP	YES	ICP is available.
...		
- ICDL	3-(4)-7	Length of DN sent to and received from the ICP.
- ICPD	(0)-9	Padding digit for DNs shorter than specified in ICDL.

LD 15 – For MMIA, enter "YES" to both the Meridian Mail prompt (IMS) and the Intercept Computer prompt (ICP).

Prompt	Response	Description
...		
TYPE:	IMS	Integrated Message Service Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
IMS	YES	Meridian Mail is available for customer number.
...	...	...
TYPE	ICP–DATA	Intercept computer update.
- ICP	YES	ICP is available for customer number.

Feature operation

No specific operating procedures are required to use this feature.

In-Band Automatic Number Identification

Contents

This section contains information on the following topics:

Feature description 549
Operating parameters 551
Feature interactions 551
Feature packaging 554
Feature implementation 554
Feature operation 555

Feature description

The In-Band Automatic Number Identification (IANI) feature provides the ability to display a ten-digit calling party number during setup (signaling) over a non-Integrated Services Digital Network (ISDN) T1 trunk. The Automatic Number Identification (ANI) digits are displayed when they auto-terminate to an Automatic Call Distribution (ACD) Directory Number (DN) agent telephone with digit display. The IANI feature supports ten digits for ANI, or three and four digits for Dialed Number Identification (DNIS). IANI sends these digits to three places: the Call Detail Recording (CDR) records, the host, and the agent telephone.

When a Direct Inward Dialing (DID) or TIE trunk originates a call, the software determines whether the call is on an IANI trunk group. If it is, the ten ANI digits are collected, and the call auto-terminates at the ACD DN specified for that trunk, provided that the ACD telephone has digit display and Standard Delayed Display (DDS) Classes of Service. The call, sent by Dual Tone Multifrequency (DTMF) signaling prior to call termination, is not received until all the digits are received by the software.

When the call is presented to the ACD DN, a PCI message is simultaneously sent across the Application Module Link (AML) carrying the ANI digits. The message contains the ANI number, the ACD DN, and the ACD Agent ID.

If an auto-terminating ACD DN is not configured for the trunk, the call intercepts to the attendant, and the ANI number is displayed on the Attendant Console. If the call is extended to an ACD DN, the IANI digits are displayed after it is extended.

Call Detail Recording (CDR) records

Because IANI and Integrated Services Digital Network (ISDN) cannot be configured on the same trunk group, the IANI report is able to appear in place of the Calling Line Identification (CLID) records. The ANI number is shown on the second line of the CDR report in the following format:

N 002 00 T00004 01 03/24 10:15 00:00:38 4155551212******

where:

N	= record type
002	= record sequence number
00	= customer number
T00004	= trunk route and member number
01	= ACD Agent Position ID
03/24	= date (month/day)
10:15	= time (hour:minute)
00:00:38	= duration (hours:minutes:seconds)
4155551212******	= ANI number (ten digits followed by ******)

For a complete description of CDR output, see *Call Detail Recording: Description and Formats* (553-3001-350).

Operating parameters

IANI operates on T1, Direct Inward Dialing (DID), and TIE trunks only.

IANI cannot be configured on the same trunk with Electronic Switched Network (ESN), Integrated Services Digital Network (ISDN), or Dialed Number Identification Service (DNIS).

The auto-terminating Automatic Call Distribution (ACD) Directory Number (DN) is configured in LD 14. Any ACD agent specified to answer IANI calls also receives standard ACD calls. When a standard ACD call is received on a non-ISDN or non-ANI trunk, no ANI numbers are displayed.

If an IANI call terminates on a non-ACD DN, no ANI digits appear on the telephone display. Likewise, no PCI messages are sent across the Application Module Link (AML).

Auxiliary Processor Link (APL) is not supported.

Should the system initialize while an agent is active on an IANI call, there will be no impact on the call. However, if any call modification (such as, Call Transfer or Conference) takes place, the ANI number is lost.

A Dual Tone Multifrequency (DTMF) receiver is required to interpret the DTMF tones with an IANI number.

Feature interactions

The IANI feature interacts a great deal with ACD. For a complete description of the ACD features involved, see *Automatic Call Distribution: Description* (553-3001-351).

ACD Answer
Call Supervisor
Emergency

If the agent presses the Supervisor (ASP) key or the Emergency (EMR) key, the digit display is cleared when the supervisor answers the call. The display remains clear while the supervisor is active on the call. If the supervisor releases first, the ANI number reappears on the agent's telephone display.

ACD Interflow

If an IANI call interflows to another predesignated local ACD DN, the ANI number is displayed on the overflow agent's digit display. The source ACD DN is displayed following the ANI number.

ACD Night Call Forward

If an ANI call is forwarded to an ACD DN, the ANI number is displayed on the ACD Agent telephone.

ACD Overflow by Count

If an IANI call overflows to another ACD DN, the ANI number is displayed on the overflow agent's digit display. The source ACD DN is displayed following the ANI number.

Activity code

If the Activity Code (ACNT) key is activated during an IANI call, the display is cleared. Once the activity code has been entered and the ACNT key pressed again, the ANI number reappears on the agent's display.

Attendant Recall

If an ACD Agent is active on an IANI call and activates the Attendant Recall (ARC) key to call the attendant, the agent's display shows the attendant number when the attendant answers the call. The ANI number reappears when the attendant releases.

Call Consultation

If the agent is active on an IANI call and presses the TRN key for call consultation, the display is cleared. When the agent restores the IANI call, the ANI number reappears.

Call Park

If an agent parks an IANI call and it times out and recalls the agent, the ANI number is not displayed.

Call Transfer

If an agent transfers an IANI call to another ACD DN, the ANI number is displayed on the terminating set's display.

Conference

If an agent activates the Conference feature while active on an IANI call, the display is cleared. The display remains clear while the Conference call is active. If the conferenced party releases first, the ANI number appears on the agent's display.

Display key

If the agent is active on an IANI call and presses the Display (DSP) key to display another key feature, the ANI number does not reappear when the DSP function is complete.

Hold

If an ACD Agent places an IANI call on hold, the ANI number reappears when the call is restored.

Network ACD

If an IANI call diverts to a target node as a result of Network ACD (NACD), the ANI number appears at the target node.

R2MFC Calling Number Identification/Call Detail Recording Enhancements

Inband ANI trunks do not support CNI. If a CNI is available in addition to the IANI on an IANI trunk, the IANI would be used for the CLID.

Time and date

If the agent presses the Time and Date (TAD) key while on an IANI call, the time and date remain displayed throughout the call. To display the ANI number again, place the call on hold and retrieve it. The ANI number reappears.

Time overflow

If an ACD Agent receives an IANI call due to time overflow, the ANI number is displayed. The source ACD DN follows the ANI number on the display.

Virtual Agents

Virtual Agents are not supported for IANI calls.

Feature packaging

The In-Band ANI (IANI) feature is not packaged separately. Implementation of IANI requires the following packages:

- Basic ACD (BACD) package 40
- ISDN Signaling (ISDN) package 145
- 1.5 Mbps Primary Rate Access (PRA) package 146
- Inter Exchange Carrier (IEC) package 149, and
- Dialed Number Identification Service (DNIS) package 98.

If Application Module Link (AML) is required, Command Status Link (CSL) package 77, and Integrated Messaging System (IMS) package 35, must be included.

For CDR records, Call Detail Recording (CDR) package 4 is required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Identify the route as an In-Band Automatic Number Identification route.

2 LD 23 – Send the IANI messages across the Auxiliary Processor Link (APL).

LD 16 – Identify the route as an In-Band Automatic Number Identification route.

Prompt	Response	Description
REQ	NEW CHG	Add or change an IANI route.
TYPE	DID TIE	Direct Inward Dialing (DID) or TIE route.
ISDN	NO YES	Enable or disable ISDN (cannot be configured on same route as IANI).
AUTO	(NO) YES	(Do not) specify as an auto-terminating route.
IANI	(NO) YES	(Disable) enable the IANI route.

LD 23 – Send the IANI messages across the Auxiliary Processor Link (APL).

Prompt	Response	Description
REQ	NEW CHG	Add or modify an IANI route.
TYPE	ACD	IANI calls terminate at an auto-terminating ACD DN.
ISAP	YES (NO)	Enable IANI messaging across the AP link.

Feature operation

No specific operating procedures are required to use this feature.

Incoming Call Indicator Enhancement

Contents

This section contains information on the following topics:

Feature description 557
Operating parameters 557
Feature interactions 558
Feature packaging 559
Feature implementation 559
Feature operation 559

Feature description

This enhancement introduces the Incoming Call Indicator (ICI) – the RDI-intercept ICI on the Attendant Console. This ICI identifies a Direct Inward Dialing (DID) call that has been intercepted to the attendant because the destination station is restricted from receiving DID calls (RDI Class of Service).

Operating parameters

If the attendant is within a system network, a special signal must be sent to the attendant when RDI-intercept to the attendant occurs.

Feature interactions

AC15 Recall: Transfer from Norstar

If the held party recalls the attendant due to intercept or recall treatment, the recall is presented to the corresponding ICI key (INT or RLL).

Attendant Recall

If an RDI-intercepted call that is extended by the attendant to the destination party having RDI Class of Service is either transferred back or recalled to the attendant, then the attendant recall ICI lights up and not the RDI-intercept ICI.

Call Forward All Calls
Call Forward Busy

When a DID call to station that is unrestricted from receiving DID calls (UDI Class of Service) is forwarded to a UDI station due to Call Forward All Calls or Call Forward Busy, the call is RDI-intercepted to the attendant. The attendant display shows the DN of the dialed party.

If the call has been forwarded to the attendant, the Call Forward All Calls/Call Forward Busy ICI lights up, and not the RDI-intercept ICI.

Call Forward No Answer

When a DID call to a station that is unrestricted from receiving DID calls (UDI Class of Service) is forwarded to a UDI station due to Call Forward No Answer, the call is not RDI-intercepted to the attendant. The dialed party continues to ring. If the call has been forwarded to the attendant, the Call Forward No Answer ICI lights up, and not the RDI-intercept ICI.

Slow Answer Recall

If an RDI-intercepted call that is extended by the attendant to the destination party having RDI Class of Service is recalled to the attendant due to Slow Answer Recall, then the Call Forward No Answer ICI lights up and not the RDI-intercept ICI. The attendant display shows the DN of the dialed party.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

LD 15 – Respond to the ICI prompt with the ICI number:

Prompt	Response	Description
...		
ICI	x RDI	ICI number; RDI intercept. x = key number (from 0 to 19).

Feature operation

When the call is intercepted to the attendant, the RDI-intercept ICI becomes lit. The attendant can then answer the call, and extend it to the destination party if desired.

Incoming DID Digit Conversion

Contents

This section contains information on the following topics:

Feature description 561
Operating parameters 564
Feature interactions 565
Feature packaging 566
Feature implementation 566
Feature operation 568

Feature description

The Incoming DID Digit Conversion (IDC) feature allows digits received from the Central Office (CO) to be converted to unrelated extension numbers within the system. This conversion is accomplished using a translation table dedicated to a Direct Inward Dialing (DID) route. The digit conversion table is set up to map the received (external) DID digits into the local (internal) Directory Number (DN).

IDC can be selectively applied to DID routes. A unique conversion table is available for each route.

Full Digit Conversion

All the digits received are converted to another string of digits as specified in the conversion table.

Different strings of digits can be converted to the same internal Directory Number (DN).

Partial Digit Conversion

Not all of the digits received from the Central Office (CO) are converted. The remaining digits may remain unchanged, and the whole string of digits is forwarded to the Directory Number (DN) translator.

It is possible to convert a partial string of digits to another partial string of digits of a different length (for example, 23xx to 4xx or 2xx to 49xx). The range of DNs to convert can include a mix of DN lengths.

No Digit Conversion

If the digits received are not defined in the conversion table, they are assumed to represent an internal Directory Number (DN). They are forwarded to the DN translator without any change.

An empty IDC table should not be programmed in a live DID route. If this is done all calls to the DID route intercept to the attendant.

Direct Call Termination

Incoming calls from non-Direct Inward Dialing (DID) trunks are not affected by Incoming DID Digit Conversion (IDC). If a call from a trunk on a route with IDC is received, the digits are translated into a pass (continue) or a converted telephone of local digits. These digits replace the dialed digits. Additional dialed digits are then forwarded directly for call processing. The IDC processor has no further influence on the call. Once the internal digit processor receives the digits, it alone determines the disposition of the call. It may be able to terminate the call, or it may be required to intercept the call due to invalid digits, a busy station, or Call Forward.

When DEXT = NO (LD 16) the Meridian 1 proprietary telephone display looks like this:

AAAA:MMM

where:

AAAA = route access code, and

MMM = Route Member Number.

The display may show the name of the route if Call Party Name Display (CPND) is allowed.

When DEXT = YES (LD 16) the Meridian 1 proprietary telephone display looks like this:

AAAA:MMM Pxxxx

where:

AAAA = route access code

MMM = Route Member Number

P = Special character (identifying the received digits), and

xxxx = Originally dialed digits (preconverted).

When DEXT = NO (LD 16) the Attendant Console display looks like this:

AAAA:MMM iiii xxxx

where:

AAAA = route access code

MMM = Route Member Number

iiii = Internal DN (called party), and

xxxx = route name if Call Party Name Display (CPND) is allowed.

When DEXT = YES (LD 16) the Attendant Console display looks like this:

AAAA:MMM#:xxxx iiii

where:

AAAA = route access code

MMM = Route Member Number

= Special character (identifying the received digits)

xxxx = originally dialed digits, and

iiii = Internal DN (called party).

Incoming Call Redirection

If an incoming call is redirected to a Centralized Attendant Services (CAS) or local attendant, the local DN is used to extend the call. If an incoming call reaches a Night DN, Hunt DN, Call Forward DN, or similar destination, then both the internal DN and the directory of local DNs are used to redirect the call.

Operating parameters

IDC applies to Direct Inward Dialing (DID) routes only. Auto-terminate trunks to Dialed Number Identification Service (DNIS) do not support IDC. All digits received from an incoming call translate to a maximum of four digits. Acceptable received digits for an incoming call are 0 through 9.

New Flexible Code Restriction (NFCR) is required to operate IDC. Since NFCR trees and IDC tables share the same structure, the total combined number of NFCR trees and IDC tables cannot exceed 255 per customer.

Feature interactions

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

An IDC table can be used to convert digits received on a DASS2 DID trunk into a digit string having the UDP format. This allows a DASS2 DID call to access the DPNSS1 UDP network.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the Direct Inward Dialing (DID) number to dial to reach the set. Therefore, if an external party uses a DN, delivered in an ANI message, for making a call to the corresponding extension, the call may fail.

EuroISDN Continuation

The Incoming Digit Conversion (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

EuroISDN Master Mode

IDC is supported on the incoming EuroISDN Master Mode connectivity DID routes. If IDC is equipped, digits received as a called party number are converted, and digit analysis is then performed on the converted digits.

ISDN QSIG Name Display

IDC trunk and name information is passed and displayed to the terminating party when no name information is received from the Direct Inward Dial (DID) trunk. The Incoming DID Digit Conversion (IDC) feature is activated, and name information is associated with the converted digit sequence.

Name information received from a DID trunk takes precedence over an IDC trunk name.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the DID number to dial to reach the set. Therefore, if an external party uses a DN for making a call to the corresponding extension which is delivered in an ANI message, the call may fail.

Feature packaging

Incoming Digit Conversion (IDC) package 113 requires New Flexible Code Restriction (NFCR) package 49.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Specify the maximum number of Incoming Digit Conversion trees allowed.
2. LD 49 – Create the IDC tables to convert incoming Direct Inward Dialing digits by specifying the IDC tree and customer numbers.
3. LD 16 – Enable the digit conversion for required DID trunk routes.

LD 15 – Specify the maximum number of Incoming Digit Conversion trees allowed.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FCR	New Flexible Code Restrictions Option.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- NFCR	(NO) YES	(Disable) enable New Flexible Code Restriction (NFCR).

- MAXT	1-255	Maximum number of NFCR trees.
- IDCA	(NO) YES	(Disable) enable IDC.
- DCMX	1-255	Maximum number of IDC tables. ***Note:*** The sum of the values for MAXT and DCMX cannot exceed 255 per customer.

LD 49 – Create the IDC tables to convert incoming Direct Inward Dialing digits by specifying the IDC tree and customer numbers.

Prompt	Response	Description
REQ	NEW	Create tables.
TYPE	IDC	IDC tables.
CUST	xx	Customer number, as defined in LD 15
DCNO	0-254	IDC tree number.
IDGT	0-9999 0-9999	DN or range of DNs to be converted. Examples: To convert the external DN 3440 to 510, enter: *PromptResponse* IDGT3440 3440510 To convert external DNs in the range 3440–3465, enter: *PromptResponse* IDGT3440 3465 3440444 3441445 — — — — — — 3465469

LD 16 – Enable the digit conversion for required DID trunk routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
IDC	YES	Use digit conversion for this route.
- DCNO	0-254	IDC tree number.
- NDNO	0-254	IDC conversion table for Night mode.
- DEXT	(NO) YES	(Do not) allow Digit Display.

Feature operation

No specific operating procedures are required to use this feature.

Incoming Digit Conversion Enhancement

Contents

This section contains information on the following topics:

Feature description 569
Operating parameters 569
Feature interactions 570
Feature packaging 571
Feature implementation 571
Feature operation 571

Feature description

The Incoming Digit Conversion (IDC) feature allows conversion into a DN of up to eight digits. The feature can operate as standalone or in an ISDN environment. The conversion is applied at the network node on which the call comes and before the digits are processed, so that there are no ISDN signaling requirements.

Operating parameters

IDC applies to Direct Inward Dialing (DID) routes only. Auto-terminate trunks to Dialed Number Identification Service (DNIS) do not support IDC. All digits received from an incoming call translate to a maximum of four digits. Acceptable received digits for an incoming call are 0 through 9.

New Flexible Code Restriction (NFCR) is required to operate IDC. Since NFCR trees and IDC tables share the same structure, the total combined number of NFCR trees and IDC tables cannot exceed 255 per customer.

Feature interactions

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the Direct Inward Dialing (DID) number to dial to reach the set. Therefore, if an external party uses a DN, delivered in an ANI message, for making a call to the corresponding extension, the call may fail.

EuroISDN Continuation

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

EuroISDN Trunk - Network Side

This feature is supported on the incoming EuroISDN Trunk - Network Side connectivity DID routes. If IDC is equipped, digits received as a called party number are converted, and digit analysis is then performed on the converted digits.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the DID number to dial to reach the set. Therefore, if an external party uses a DN for making a call to the corresponding extension which is delivered in an ANI message, the call may fail.

Feature packaging

Incoming Digit Conversion Enhancement is included in Incoming Digit Conversion (IDC) package 113 that requires New Flexible Code Restriction (NFCR) package 49.

Feature implementation

To implement Incoming Digit Conversion Enhancement, see "Incoming DID Digit Conversion" on page 561 in this document.

Feature operation

No specific operating procedures are required to use this feature.

Incremental Software Management

Contents

This section contains information on the following topics:

Feature description 573
Operating parameters 593
Feature interactions 593
Feature packaging 593
Feature implementation 595
Feature operation 595

Feature description

Incremental Software Management (ISM) is a feature that provides flexibility and control over system configuration and implementation. With ISM, software ordering and pricing is based on the total count of used ISM parameters.

With the introduction of Succession 3.0 software, ISM has been organized into the following three categories:

- Service ISM
- System ISM
- Pre-set ISM

ISM allows key codes to control the increments provided for Service- and System-type ISM when filling customer orders. The ISM increments are the same globally and are common to all system types. Some regions do not use certain ISM parameters. For example, North America does not use Wireless Visitors. If this ISM parameter is introduced to North America, it will be in accordance with Global Software Structure.

Service ISM

Service ISM parameters are chargeable. They are aligned with the five service levels.

The Customer/Distributors must first select the service level and then the number of Service ISM parameters for each type. All Service ISM parameters must be ordered and filled at the same service level. No combinations or mixing of ISM parameters with other levels are allowed. For example, do not mix Level 1 analog ISMs parameters with a customer who is at level 2 Enhanced Business. This customer must be serviced with all ISM parameters at the level 2 Enhanced Business level.

Service ISM parameters use the same Global code for all systems and regional service levels. They are assigned incremental defaults, minimum order quantity amounts, and incremental amounts.

Service ISM parameters, when ordered, are used to indicate the software service level the customer wishes to be operating on. For example, when ISM parameters are ordered for level 3B, the order management systems automatically provide the customer this service level with the appropriate regional software feature package content.

When allowable limits are exceeded, any additional entry is blocked, and an error message is shown every time a subsequent entry is attempted.

System ISM

System ISM parameters are chargeable. They are applicable to the complete system and are not dependent on which service level the customer is using.

Note: The same order code is used for all system types and regions.

Pre-set ISM

Pre-set ISM parameters are preset in the Nortel Networks factory with their increments set to the maximum. These are non-chargeable but do provide value to the customer. Setting these ISM parameters to the maximum allows customers to configure ISM parameters to meet their feature and configuration needs.

> ***Note:*** TNs are not used to control user capacity, but are set to the maximum amount for all system types.

Table 13 lists the ISM parameters available with Succession 3.0 Software.

Table 13
ISM parameters with Succession 3.0 Software

Service ISMs	System ISM parameters	Pre-set ISM parameters
Analogue Telephones	Personal Call Assistant	TNs
CLASS Telephones	ITG ISDN Trunks	ACDN
Digital Telephones	IP PEER H.323 Trunks	AML
Wireless Telephones	AST	BRAND
IP Phones	RAN CON	LTID
Wireless Visitors	MUS CON	RAN RTE
ACD Agents	Survivability	Attendant Consoles
		BRI DSL
		MPH DSL
		DATA Ports
		Phantom Ports
		Traditional Trunks
		DCH
		TMDI D-Channels

Table 14 lists ISM dependencies for TN configurations.

Table 14
ISM parameters dependencies on TN configuration (Part 1 of 5)

ISM mnemonic	How a TN is configured
Service ISM parameters	
Analogue Telephones	This parameter counts analog (500/2500-type) telephones configured in LD 10, including: • analog ACD agents and AST • Line-side T1/E1 devices, such as voice mail systems, voice response units, and trading turrets (LD 10, TYPE 500) • Faxes and modems (LD 10, TYPE 500, CLS FAXA) • Fax Server ports (LD 10, TYPE 500, FTR FAXS) Phantom ports, wireless and CLASS sets are not counted (LD 10, TYPE 500, WRLS is NO, CLS CNUD and CNAD).
CLASS Telephones	This parameter counts CLASS compatible analog (500/2500-type) sets (LD 10, TYPE 500, CLS CNAA or CNUA).
Digital Telephones	This parameter counts digital sets, including digital ACD agents and AST. Meridian Mail/Call Pilot ports, data and phantom ports, and Virtual sets are not counted.
Wireless Telephones	This parameter will count Companion and Companion DECT sets (LD 10, TYPE 500, WRLS YES) DECT sets supporting concentration. Visiting DECT sets are not counted.
Internet Telephones	This parameter counts Internet sets (LD 11, TYPE i2002 or i2004) and Softphone (LD 11, TYPE i2050).
Wireless Visitors	This parameter counts Visiting DECT sets supporting concentration feature (LD 10, TYPE DCS, VSIT YES).

Table 14
ISM parameters dependencies on TN configuration (Part 2 of 5)

ISM mnemonic	How a TN is configured
ACD Agents	This parameter counts Analog ACD agents (LD 10, TYPE 500, CLS AGTA, FTR ACD), Wireless ACD agents (LD 10, TYPE DCS or 500, WRLS YES, CLS AGTA, FTR ACD), Digital ACD agents, Meridian Integrated ACD ports, Virtual Office host agents, and Internet ACD agents (LD 11, TYPE i2002, i2004, i2050, KEY 0 ACD). Meridian Mail/Call Pilot ports are not counted.
System ISM parameters	
Personal Call Assistants	This parameter counts Personal Call Assistant data blocks (LD 11, TYPE PCA, KEY 1 HOT P).
ITG ISDN trunks	This parameter counts ITG-i486 Card trunks, ITG-Pentium Card trunks, and ITG Media Card trunks. Voice gateways are not counted (LD 14, TYPE not VGW, XTRK ITG8, ITGP, MC8, MC32 and not VGW, IPTN NO).
IP Peer H.323 trunks	This parameter counts Virtual trunks (LD 14, TYPE TIE, XTRK VTRK).
AST	This parameter counts Associated analog sets (LD 10, TYPE 500, AST YES), Associated analog ACD agents (LD 10, TYPE 500, CLS AGTA, FTR ACD, AACD YES), Associated digital and internet sets, and Associated trunks. The following trunks cannot be associated: MUS, ADM, R232, R422, MCU, MDM, AWR, PAG, DIC, RAN, RCD) Meridian Mail/Call Pilot ports are not counted.
RAN connections	This parameter counts Broadcasting RAN trunks (LD 14, TYPE RAN).
Music connections	This parameter counts Broadcasting music connections. Non-broadcasting music trunks are not counted (LD 14, TYPE MUS).1 Music Broadcasting trunk = 64 Music Connections.

Table 14
ISM parameters dependencies on TN configuration (Part 3 of 5)

ISM mnemonic	How a TN is configured
Survivability	This parameter counts Survivability ISM (How many expansion cabinets/Succession Media Gateways can operate in survivable mode). This parameter is specific to Small Systems and Succession 1000 systems. (LD 117, SURV cab YES).
Pre-set ISM parameters	
TNs	The total number of TNs refers to Terminal Numbers (TNs) configured in LDs 10, 11, 12, 13, and 14. There is no differentiation among signaling, data, and voice channels.
ACD DNs	ACD DNs counts the number of ACD and CDN data blocks (LD 23, TYPE ACD or CDN).
AMLs	Application Module Links (LD 17, ADAN AML).
Brand	Brand index ISM specifies a string displayed on an idle set.
LTIDs	Logical terminals configured on DSLs (LD 27, TYPE DSL).
RAN routes	Recorded Announcement Routes (LD 16, TKTP RAN).
Attendant consoles	This parameter counts every attendant console and PC console configured in LD 12. An attendant console can use two or more TNs. However, the number of TNs occupied by an attendant console are not used for attendant console ISM counting criteria; each TN occupied by an attendant console is used for System TN ISM counting criteria. TNs used for power supply are not counted toward attendant consoles.
BRI DSLs	This parameter counts every BRI line (LD 27, TYPE DSL, APPL BRIL).
MPH DSLs	This parameter counts every BRI MPH line (LD 27, TYPE DSL, APPL MPH).

Table 14
ISM parameters dependencies on TN configuration (Part 4 of 5)

ISM mnemonic	How a TN is configured
Data ports	This parameter counts every Data Port configured in LDs 10 (data TNs), 11 (data TNs) or 14 (MCA, MCU). Data Ports are excluded from counting as Analogue or Digital Telephones or Traditional Trunks. A data TN configured in LD 11 is a Data Port. A Meridian Communications Adapter (MCA) fits inside a Meridian Digital Telephone to provide access to data functions. An MCA is configured in LD 11 as an M2006, M2008, M2216 or M2616 or M3900 series with DTAO prompt set to either Meridian Programmable Data Adapter (MPDA) or MCA. A Meridian Communications Unit (MCU) that replicates the functionality of the MCA and provides additional features. Both MCA and MCU are counted as Data Ports. A Data Access Card (DAC) is a data interface card that allows the card to work with the RS-232 interface, the RS-422 interface, or both. Configuration of DAC is in LD 11, with R232 or R422 as the TYPE prompt. Both R232 and R422 data sets are counted as Data Ports. A Data Port is not limited to units 16–31. If a TN has Flexible Voice/Data Allowed (FLXA) CLS, a DATA port is allowed to the TN (unit 0-15). ATA sets (LD 11, TYPE any of M3000, CLS DTA and not MMA) Meridian Communication Adapters (LD 11, TYPE any of M2000, CLS DTA and not MMA) Meridian Communication Units (LD 11, TYPE MCU)R232 DAC units (LD 14, TYPE R232) R422 DAC units (LD 14, TYPE R422).
Phantom ports	Analog phantom sets configured on Phantom loops (LD 10, TYPE 500) Digital phantom ports configured on Phantom loops (LD 11, TYPE any of M2000 and M3000)

Table 14
ISM parameters dependencies on TN configuration (Part 5 of 5)

ISM mnemonic	How a TN is configured
Traditional Trunks	This parameter counts each Traditional Trunk (analog, digital, ISDN, and ITG 1.0 Trunks) configured in LD 14. Analog trunks that use in-band signaling for establishing calls to COs or other switches are counted as Traditional Trunks. Trunks of this nature include, but are not limited to, the following: • Automatic Identification of Outward Dial (AIOD) • Common Control Switching Arrangement (CCSA) • Automatic Number Identification (ANI) • Autovon (ATVN) • Central Automatic Message Accounting (CAMA) • Central Office (COT) • Common Control Switching arrangement (CSA) • Direct Inward Dial (DID) • Foreign Exchange (FEX) • Feature Group D (FGD) • Release Link Main (RLM) • Release Link Remote (RLR) • TIE • Wide Area Telephone Service (WAT) Counting analog trunks does not depend on hardware type, density or country-specificity. DTI channels (1.5 and 2.0 Mb) and JDMI trunks count as Traditional Trunks. Line-Side T1/E1 are counted as Analogue Telephones and are not counted as Traditional Trunks. ISDN trunks such as ISL, VNS, 1.5 and 2.0 Mb PRI (including IDA) and BRI count as Traditional Trunks.
D-channels	Primary D-channels (LD 17, ADAN DCH) Backup primary D-channels (LD 17, ADAN BDCH)
TMDI D-channels	D-channels configured on the TMDI card. This parameter is specific to Small Systems and Succession 1000 systems (LD 17, TYPE ADAN, CTYP TMDI).

Table 15 lists Service Level ISM default and increment values.

Table 15
Service Level ISM default and increment values (Part 1 of 2)

ISM mnemonic	New system default setting by system type	Order increment for new systems and expansions	ISM ordering guidelines
Analog user ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C 8 - 1000 8 - 1000 M Cabinet/ Chassis 8 - 1000 M-HG/SG/ MG	
Class user ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C 8 - 1000 8 - 1000 M Cabinet/ Chassis 8 - 1000 M-HG/SG/ MG	
Digital user ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C 8 - 1000 8 - 1000 M Cabinet/ Chassis 8 - 1000 M-HG/SG/ MG	Provision 8 for mini CallPilot
Wireless user ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C 8 - 1000 8 - 1000 M Cabinet/ Chassis 8 - 1000 M-HG/SG/ MG	See Note 2 below.

Table 15
Service Level ISM default and increment values (Part 2 of 2)

ISM mnemonic	New system default setting by system type	Order increment for new systems and expansions	ISM ordering guidelines
IP user ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C 8 - 1000 8 - 1000 M Cabinet/ Chassis 8 - 1000 M-HG/SG/ MG	
Wireless Visitor user ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C 8 - 1000 8 - 1000 M Cabinet/ Chassis 8 - 1000 M-HG/SG/ MG	This ISM parameter is only used in EMEA and Asia Pacific Regions
ACD Agents user ISM	10 - Option 11C Cabinet/Chassis 10 - Option 61C/81C 10 - 1000 10 - 1000 M Cabinet/ Chassis 10 - 1000 M-HG/SG/ MG	1 - Option 11C Cabinet/Chassis 1 - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis 1- 1000 M-HG/SG/ MG	

Note 1: "Option 61C/81C" refers to Meridian 1 Option 61C CP PII and Meridian 1 Option 81C CP PII. "1000 M-HG/SG/MG" refers to Succession 1000M Half Group, Succession 1000M Single Group, and Succession 1000M Multi Group.

Note 2: For North America and CALA, Wireless ISM parameters are used for upgrades, expansions, and transfers. This ISM is not supported on North American Succession 1000 systems and Branch Office. North American and CALA Companion Wireless ISMs can be moved from an existing system to the Succession 3.0 Software structure, provided a customer provides a LD 22 or Order Pro report indicating the number of Users that need to be transferred to the new system using the Succession 3.0 Software structure.

Table 16 lists System Level ISM default and increment values.

Table 16
System Level ISM default and increment values (Part 1 of 2)

ISM mnemonic	New system default setting by system type	Order increment for new systems and expansions	ISM ordering guidelines
Personal Call Assistant ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	1 - Option 11C Cabinet/Chassis 1 - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis 1 - 1000 M-HG/SG/ MG	These increments apply to standalone Succession 1000M, Succession 1000, and Meridian 1 systems.
ITG ISDN Trunk ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C N/A - 1000 N/A - 1000 M Cabinet/ Chassis N/A - 1000 M-HG/SG/ MG	8 - Option 11C Cabinet/Chassis 8 - Option 61C/81C N/A - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	ITG ISDN Trunk can be co-resident in a configuration with IP Peer H323 Trunk for Large Systems.
IP Peer H323 Trunk ISM	N/A - Option 11C Cabinet/Chassis N/A - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	N/A - Option 11C Cabinet/Chassis N/A - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis 1 - 1000 M-HG/SG/ MG	New ISM for Succession 3.0. Also provides Virtual Trunk ISM for Succession 1000 systems.
AST ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	1 - Option 11C Cabinet/Chassis 1 - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis 1 - 1000 M-HG/SG/ MG	This ISM controls Nortel and third-party applications.

Table 16
System Level ISM default and increment values (Part 2 of 2)

ISM mnemonic	New system default setting by system type	Order increment for new systems and expansions	ISM ordering guidelines
RAN CON ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	1 - Option 11C Cabinet/Chassis 1 - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis 1 - 1000 M-HG/SG/ MG	
MUS CON ISM	0 - Option 11C Cabinet/Chassis 0 - Option 61C/81C 0 - 1000 0 - 1000 M Cabinet/ Chassis 0 - 1000 M-HG/SG/ MG	1 - Option 11C Cabinet/Chassis 1 - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis 1 - 1000 M-HG/SG/ MG	
Survivability ISM	0 - Option 11C Cabinet/Chassis N/A - Option 61C/81C 1 - 1000 0 - 1000 M Cabinet/ Chassis N/A - 1000 M-HG/SG/ MG	1 - Option 11C Cabinet/Chassis N/A - Option 61C/81C 1 - 1000 1 - 1000 M Cabinet/ Chassis N/A - 1000 M-HG/SG/ MG	

Table 17 lists factory pre-set ISM values.

Table 17
Factory pre-set ISM values (Part 1 of 3)

ISM mnemonic	Value setting by system type
TNS	2500 - Option 11C Cabinet/Chassis 32760 - Option 61C/81C 2500 - 1000 2500 - 1000 M Cabinet/Chassis 32760 - 1000 M-HG/SG/MG
AML	N/A - Option 11C Cabinet/Chassis 16 - Option 61C/81C N/A - 1000 N/A - 1000 M Cabinet/Chassis 16 - 1000 M-HG/SG/MG
LTID	0 - Option 11C Cabinet/Chassis 32760 - Option 61C/81C N/A - 1000 N/A - 1000 M Cabinet/Chassis 32760 - 1000 M-HG/SG/MG
ATTENDANT CONSOLES	2500 - Option 11C Cabinet/Chassis 32760 - Option 61C/81C 2500 - 1000 2500 - 1000 M Cabinet/Chassis 32760 - 1000 M-HG/SG/MG
MPH DSL	N/A - Option 11C Cabinet/Chassis 64 - Option 61C/81C N/A - 1000 N/A - 1000 M Cabinet/Chassis 64 - 1000 M-HG/SG/MG
PHANTOM PORTS	2500 - Option 11C Cabinet/Chassis 32760 - Option 61C/81C 2500 - 1000 2500 - 1000 M Cabinet/Chassis 32760 - 1000 M-HG/SG/MG

Table 17
Factory pre-set ISM values (Part 2 of 3)

ISM mnemonic	Value setting by system type
DCH	N/A - Option 11C Cabinet/Chassis 254 - Option 61C/81C N/A - 1000 N/A - 1000 M Cabinet/Chassis 254 - 1000M-HG/SG/MG
ACDN	300 - Option 11C Cabinet/Chassis 24000 - Option 61C/81C 300 - 1000 300 - 1000 M Cabinet/Chassis 24000 - 1000 M-HG/SG/MG
BRAND	2 - Option 11C Cabinet/Chassis 2 - Option 61C/81C 2 - 1000 2 - 1000 M Cabinet/Chassis 2 - 1000 M-HG/SG/MG
RAN RTE	500 - Option 11C Cabinet/Chassis 512 - Option 61C/81C 500 - 1000 500 - 1000 M Cabinet/Chassis 512 - 1000 M-HG/SG/MG
BRI DSL	150 - Option 11C Cabinet/Chassis 10000 - Option 61C/81C 150 - 1000 150 - 1000 M Cabinet/Chassis 10000 -1000 M-HG/SG/MG
DATA PORTS	2500 - Option 11C Cabinet/Chassis 32760 - Option 61C/81C 2500 - 1000 2500 - 1000 M Cabinet/Chassis 32760 -1000 M-HG/SG/MG

Table 17
Factory pre-set ISM values (Part 3 of 3)

ISM mnemonic	Value setting by system type
TRADITIONAL TRUNKS	2500 - Option 11C Cabinet/Chassis 32760 - Option 61C/81C 2500 - 1000 2500 -1000 M Cabinet/Chassis 32760 -1000 M-HG/SG/MG
TMDI D - CHANNELS	64 - Option 11C Cabinet/Chassis N/A - Option 61C/81C 64 - 1000 64 - 1000 M Cabinet/Chassis N/A - 1000 M-HG/SG/MG

Table 18 lists maximum ISM limits. These are individual ISM limits, not engineering limits or rules.

Table 18
Maximum ISM limits (Part 1 of 3)

ISMs	Small Systems	Succession 1000 systems	Large Systems
Service ISM			
Analog user ISM	2 500	2 500	32 760
Class user ISM	2 500	2 500	32 760
Digital user ISM	2 500	2 500	32 760
IP user ISM	1 000	1 000	32 760
Wireless user ISM	2 500	2 500	32 760
Wireless Visitor ISM	2 500	2 500	10 000
ACD Agent user ISM	1 000	1 000	32 760

Table 18
Maximum ISM limits (Part 2 of 3)

ISMs	Small Systems	Succession 1000 systems	Large Systems
System ISM			
ITG ISDN Trunks ISM	2 500	N/A	32 760
IP Peer H323 Trunks ISM	2 500	764	32 760
Personal Call Assistant ISM	1 248	1 248	32 760
AST ISM	1 000	1 000	32 760
RAN_CON ISM	1 000	1 000	32 760
MUS_CON ISM	1 000	1 000	10 000
Survivability ISM	4	4	N/A
Pre Set ISM			
TNS	2 500	2 500	32 760
ACDN	300	300	24 000
AML	N/A	N/A	16
LTID	2 500	2 500	32 760
RAN_RTE	500	500	512
BRAND	2	2	2
Attendant Consoles	2 500	2 500	32 760
BRI_DSL	150	150	32 760
MPH_DSL	N/A	N/A	32 760
Data Ports	2 500	2 500	32 760
Phantom Ports	2 500	2 500	32 760

Table 18
Maximum ISM limits (Part 3 of 3)

ISMs	Small Systems	Succession 1000 systems	Large Systems
Traditional Trunks	2 500	2 500	32 760
DCH	N/A	N/A	255
TMDI_D-Channels	64	64	N/A

System monitoring

To assist in monitoring system growth, each time an overlay is used, a header appears in the affected overlay, reflecting the system status. The header indicates the total, available, and used quantities of the ISM parameters corresponding to the data blocks that are configured in the overlay. The counts are updated each time system activity adds or deletes one of the tracked items. When the limits are exceeded, an error message appears.

ACD parameters are preset for each system. The numbers in the header are not necessarily real limits and are subject to system configuration. Contact your Nortel Networks representative for information regarding your system limits.

A header, reflecting ISM parameters, is added to the following overlays:

- LD 10: analog (500/2500 type) telephones, CLASS telephones, wireless (500/DCS) telephones, wireless visitors, ACD agents, AST, TNs, data ports and phantom ports.
- LD 11: Meridian 1 proprietary telephones, internet telephones, ACD agents, PCAs, AST, TNs and data ports.
- LD 12: Attendant Consoles and the number of TNs.
- LD 13: Digitone receivers and tone detectors
- LD 14: AST, ITG ISDN trunks, IP Peer H.323 trunks, RAN and MUS connections, TNs, data ports and traditional trunks.
- LD 16: RAN routes

- LD 17: D-channels (DCH and TMDI DCH) and Application Module Links (AMLs)
- LD 23: ACD-DNs
- LD 27: TNs, Digital Subscriber Loops (DSLs) and Logical Terminal Identifiers (LTIDs).
- LD 117: Survivability (Small Systems and Succession 1000 systems only).

Printing ISM system limits

When REQ is set to SLT in LD 22, ISM system limits are printed. You can update the value of ISM limits either through sysload or the Instant ISM feature. You can print the new ISM limits through LD 22 after the update is complete.

The following shows the LD 22 implementation for printing system limits.

LD 22 – Print system limits.

Prompt	Response	Description
REQ	SLT	Print System Limits: Incremental Software Management.

The following is an example of a LD 22 print out for a Large System, when REQ = SLT.

ANALOGUE TELEPHONES	1160	LEFT	1017	USED	143
CLASS TELEPHONES	16	LEFT	4	USED	12
DIGITAL TELEPHONES	2520	LEFT	1866	USED	654
WIRELESS TELEPHONES	96	LEFT	96	USED	0
INTERNET TELEPHONES	1000	LEFT	782	USED	218
WIRELESS VISITORS	0	LEFT	0	USED	0
ACD AGENTS	1000	LEFT	577	USED	423
PCA	1000	LEFT	996	USED	4
ITG ISDN TRUNKS	1000	LEFT	928	USED	72
IP PEER H.323 TRUNKS	1000	LEFT	968	USED	32
AST	1000	LEFT	767	USED	233
RAN CON	0	LEFT	0	USED	0
MUS CON	0	LEFT	0	USED	0
TNS	32760	LEFT	29621	USED	3139
ACDN	24000	LEFT	23769	USED	231
AML	16	LEFT	12	USED	4
IDLE_SET_DISPLAY NORTEL					
LTID	96	LEFT	96	USED	0
RAN RTE	512	LEFT	512	USED	0
ATTENDANT CONSOLES	32760	LEFT	32760	USED	0
BRI DSL	50	LEFT	38	USED	12
DATA PORTS	32760	LEFT	32597	USED	163
PHANTOM PORTS	32760	LEFT	31986	USED	774
TRADITIONAL TRUNKS	32760	LEFT	32068	USED	692
DCH	255	LEFT	239	USED	16

System administration

When the predefined ISM limits are reached, an error message indicates that further database additions are blocked.

New keycode must be ordered to increase system limits. In order to minimize delays in system administration, it is critical that the configuration limits be monitored and that new disks be ordered before the current parameters are exceeded.

Software Upgrade

When performing a system upgrade, if any of the new ISM limits exceed present limits, then do not attempt to sysload. Excess information will be lost. Obtain new disks with expanded limits.

CAUTION

System information will be lost. Upon software upgrade, if SYS message 4327, 4328, 4329, or 4330 appears at SYSLOAD, then SYSLOAD previous system disks. Order ISM disks with sufficient system parameters configured.

DO NOT DATADUMP; system information will be lost. Call your technical support department for assistance.

Keycodes

Small System and Input-Output Disk Unit with CD-ROM (IODU/C) customers can modify ISM limits using a keycode.

A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects ISM parameters.

In order for Small System and IODU/C customers to expand ISM limits, they must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in LD 143.

To make the expansion effective, the customer must sysload. For further information on keycode installation, please refer to *Large System: Upgrade Procedures* (553-3021-258).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Incremental Software Management requires the following packages:

- ACD-DNs and ACD Agent
  - Basic ACD (BACD) package 40
- Application Module Link (AML)
  - Digit Display (DDSP) package 19
  - ACD Package B (ACD-B) package 41
  - ACD Package A (ACD-A) package 45
  - Command Status Link package 77
  - ISDN Application Module Link for Third Party Vendors (IAP3P) package 153
- AST
  - Command Status Link (CSL) package 77
  - Application Module Link (AML) package 209
- Attendant Consoles
  - Attendant Consoles is included in base system software.
- Class Telephones
  - Calling Number Delivery (CNUMB) package 332 or

— Calling Name Delivery (CNAME) package 333

- Data Ports
  - — Package requirements for data ports vary depending on the type of data port configured. Refer to *Software Input/Output: Administration* (553-3001-311) and *Software Input/Output: Maintenance* (553-3001-511) for information on specific data port package requirements.
- Internet Telephones
  - — M2000 Digital Set (DSET) package 88
  - — Aries Digital Set (ARIE) package 170
- ITG ISDN Trunks
  - — Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58
  - — Integrated Services Digital Network (ISDN) package 145
  - — ISDN Signaling Link (ISL) package 147
  - — Multi-purpose Serial Data Link (MSDL) package 222 (for Large Systems only)
- Phantom Ports
  - — Phantom TN (PHTN) package 254
- Traditional Trunks
  - — Package requirements for Traditional Trunks vary depending on the type of trunk configured. Refer to *Software Input/Output: Administration* (553-3001-311) and *Software Input/Output: Maintenance* (553-3001-511) for information on specific trunk package requirements.
- Wireless
  - — Meridian 1 Companion Option (MCMO) package 240
- Personal Call Assistant (PCA)
  - — Personal Call Assistant (PCA) package 398

- IP Peer H.323 Trunks
  - IP Peer Networking package 399

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Information Notification Service for Japan

Contents

This section contains information on the following topics:

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Operating parameters 601
Feature interactions 601
Feature packaging 603
Feature implementation 603
Feature operation 604

Feature description

The Information Notification Service for Japan (INS-J) feature allows a Japan local exchange to extract the calling line identification information received on Japan analog trunks (JCO/JDID) and to deliver it to subscribers' terminals/ trunks with display capability and customer oriented applications. In Japan, this service has already been available on ISDN. However, analog trunks are still seen as efficient alternatives to ISDN.

The INS-J feature has its own circuit card, the NT5D39 DXUT-J card. The DXUT-J is a Digital Signaling Processor-based Extended Universal Trunk card for the Japan market. The DXUT-J collects the FSK-format INS-J information sent by the CO and sends it to the system software. The DXUT-J also supports the Busy Tone Detection for Japan that is available on the EXUT-J card.

On an incoming call with INS-J, the system extracts information such as: Calling Party Number, Calling Party Name, Called Party Number, Date and Time, and, if applicable, Reason for absence of Calling Party Number/Calling Party Name. This information is passed on to the terminating party, which can be:

- a trunk
- a terminal or
- an application.

The INS-J information is sent by the CO in Frequency Shifted Key (FSK) format. The NT5D39 DXUT-J card decodes this information and sends it to the system software using SSD messages.

The system software extracts the Calling Party Number, Called Party Number, Calling Party Name, and Date and Time information, and the call termination follows the existing procedure. For example, if the call is from an incoming CO trunk, it terminates at the attendant or where designated by the system's database; if the call is a DID call, the system software extracts the information from the INS-J and terminates the call accordingly.

The INS-J information is passed on to the terminating party, which can be:

Trunks

- ISDN
  - — PRI/BRI
- R2MFC
  - — DTI/DTI2
  - — Analog

Terminals

- Digital sets
  - — SL-1
  - — Meridian Digital telephones
  - — BRIL sets

- Attendant Console

Applications

- Meridian Mail
- Meridian Link
- Meridian IVR
- Customer Controlled Routing
- Symposium Call Center Server

Call Detail Recording (CDR)

The INS-J feature is enabled and disabled on a per unit basis using a class of service in LD 14.

Figure 14 shows the operation of the INS-J feature.

Figure 14
System with INS-J feature operating

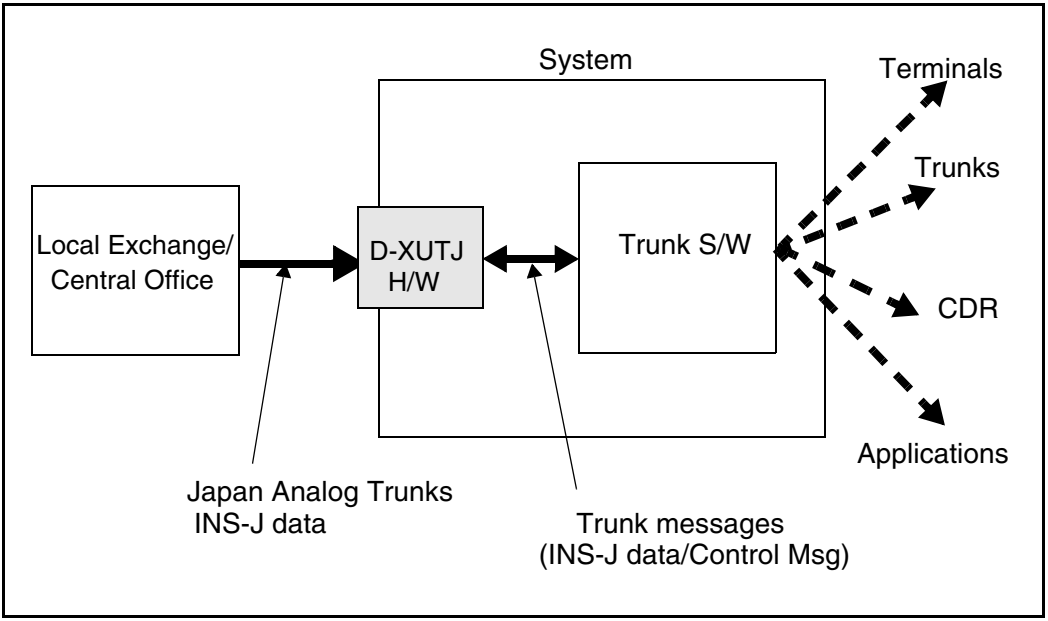
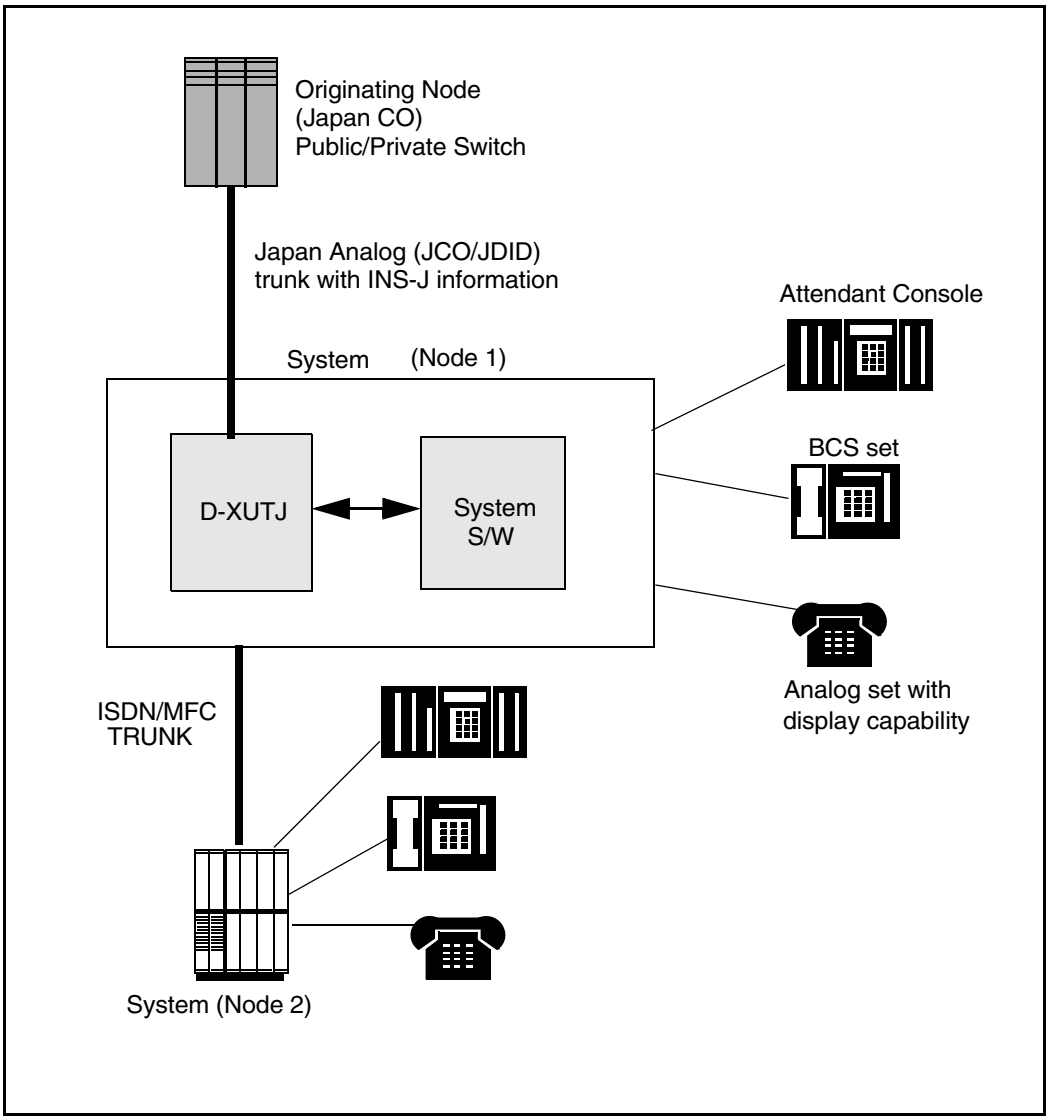


Figure 15 shows the system composition required for the INS-J CLID delivery.

Figure 15
System composition for INS-J CLID delivery



Operating parameters

This feature is only applicable for incoming analog trunks. If the terminating set/trunk cannot receive the information, then the Analog CLI information will not be displayed nor transmitted.

As per existing system functionality, only the first 16 digits of the Calling Party/Called Party number will be processed.

Display of Katakana characters is not supported. Any Katakana characters received will be ignored.

If system initialization occurs while the INS-J information is being sent from the NT5D39 DXUT-J card to the system software, then any INS-J information that has not been sent is lost and the call is lost as well, since it is not an established call. In the case of an established call, the call will be rebuilt and the display may or may not be maintained.

The system administrator must ensure that the INS-J function is activated for those trunk ports that are actually connected to a CO with INS-J.

Feature interactions

Attendant Call Extension

When an attendant extends a call from an incoming INS-J trunk, the Analog CLI information is delivered to the terminating set.

Call Transfer/Blind Transfer

When a set completes a Transfer/Blind Transfer of an incoming INS-J call, the Analog CLI information is delivered to the terminating set.

Call Forward All Calls/Call Forward No Answer/Internal Call Forward/Hunt

When a call is redirected using Call Forward All Calls/Call Forward No Answer/Internal Call Forward/Hunt, the Analog CLI information is delivered to the terminating set.

CLASS

If the call terminates on a CLASS set then the Analog CLI information is passed to the CLASS feature.

Conference/No Hold Conference

When a set receives an incoming call and then initiates a conference call, the information of the initiating set will be delivered to the terminating set, and not the Analog CLI information.

Direct Inward System Access

If a user enters the system through DISA dialing, the information passed on is that of the incoming trunk and not of the DISA DN.

Private Line Service

Private Line Service will not affect the CLI information on the set.

Basic Rate Interface (BRI)

If an incoming call from an INS-J trunk is redirected to BRI, the Analog CLI information is mapped onto the setup message and sent, as per existing system operation.

Feature Group D (FGD)

If an incoming call from an INS-J trunk is redirected to a Feature Group D trunk, the Analog CLI information is passed on as per existing system operation.

Integrated Services Digital Network (ISDN)

If an incoming call from an INS-J trunk is redirected to an ISDN trunk, the Analog CLI information is passed on as per existing system operation.

Multifrequency Compelled Signaling (MFC)

If an incoming call from an INS-J trunk is redirected to an MFC trunk, the Calling Party Number information is mapped to the CNI digits of MFC. Since MFC does not support Calling Party Name and Date/Time, that information is not sent.

Feature packaging

This feature introduces Analog CLI (ACLI) package 349.

The ACLI package requires Japan package 97.

The UK package (package 190) is incompatible with ACLI, and should not be packaged if ACLI is turned on.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Configure the Analog CLI Class of Service on a port-by-port basis.

2 LD 16 – Configure the new ring validation timer.

LD 14 – Configure the Analog CLI Class of Service on a port-by-port basis.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	COT DID	Central Office or Direct Inward Dialing.
XTRK	EXUT	Type of trunk card.
CUST	xx	Customer number, as defined in LD 15
...	...	
SUPN	YES	Supervision required.
STYP	JCO JDID	Japan CO or Japan DID.
	BTS	Busy Tone Supervision (Optional)
CLS	(CLID) CLIA	Calling Line Identification denied or allowed.
...	...	

LD 16 – Configure the new ring validation timer.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
...	...	
TKTP	TIE COT	TIE or Central Office trunk.
...	...	
CNTL	YES	Changes to controls of timers.
TIMR	RGV 256	Ring validation timer to be changed to 256.
...	...	

Feature operation

No specific operating procedures are required to use this feature.

Instant ISM

Contents

This section contains information on the following topics:

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Feature description

The Incremental Software Management limits determine the maximum number of TNs, ACD positions, and other parameters on the system.

The Instant Incremental Software Management (IISM) feature allows ISM limits to be upgraded on the system by delivering the keycode to the system, without the need for a Sysload.

During keycode activation using the existing prompts in LD 143, if the keycode is eligible for instant activation (that is, ISM parameters are the only parameters that have changed relative to the current system keycode, and no ISM limits are decreasing), the ISM limits will be upgraded “instantly.” Following successful activation, a system message introduced by the Instant ISM feature will be displayed. This message indicates that the keycode was accepted, ISM limits were increased, and that a Sysload is *not* required.

A keycode that is eligible for instant activation has ISM limits that are either unchanged or increased, has no addition or removal of feature packages, and has no changes to software release and issue, software generic, or AUX-ID.

If a keycode is not eligible for instant activation (that is, ISM parameters are lowered, or software packages are changed), system message CCBR009 ("New keycode accepted. It will be activated during the next restart.") is displayed and a Sysload *will* be required.

Operating parameters

The system does not treat the Small System MOPT parameter as an ISM limit, but rather as a package. The Instant ISM feature does not support instant MOPT changes. If the MOPT parameter is changed, a Sysload will be required.

System initialization

If system initialization occurs while a new keycode is being instantly activated, the system software will attempt to complete the keycode activation if at all possible. However, depending on when the initialization occurred, the software may not be able to complete keycode activation.

After the system has completed initialization, the craftsperson should print the active ISM parameters using LD 22. If the printed ISM parameters match the new keycode parameters, then the system software completed the keycode activation successfully. If the ISM parameters printed are the pre-upgrade parameters:

- For Large Systems, the craftsperson should access LD 143 to verify whether the new keycode is still on the hard drive by using the "KSHO HD" command. If the new keycode is still on the hard drive, then the craftsperson needs to remove the keycode from the hard drive using the "KOUT" command, and then perform the new keycode installation process again. If the new keycode is not on the hard drive, the craftsperson should perform the new keycode installation process in LD 143.
- For Small Systems and Succession 1000 systems, the craftsperson should access LD 143 and reprogram the system upgrade process.

Feature interactions

Incremental Software Management

Instant ISM does not change the operation of the various ISM limits. Instant ISM simply allows the user to upgrade ISM limits without having to Sysload.

IS-41 Networking

Instant ISM supports the MOB ISM parameter in the IS-41 Networking feature.

RAN and Music Broadcast

Certain traffic reports peg the number of times the RAN and Music ISM limits had been reached. Due to the fact that ISM limits may change instantly (without a Sysload), a traffic report that is counting the ISM hits over a period may be checking against two different values consecutively. Therefore, for one single calculation period the report will have an aberration.

Electronic Brand Line (EBLN/BRAND)

Unlike other ISM parameters which define the maximum configuration limits for various resources, the BRAND ISM parameter defines which Electronic Brand Line feature option the system is allowed to use.

The same limitation applies to the BRAND parameter as applies to other ISM limits, that is, the BRAND parameter must be unchanged or increased if the ISM limits are to be updated instantly without the need for a Sysload.

Once the BRAND ISM parameter has been increased, the user still has to access LD 17, in order to configure the actual string that is to be displayed, as per existing operation.

Telephone displays that display brandline information (when in an idle state) will not have the brandline updated immediately. The update will occur on a set the next time LAMPAUDIT audits the set.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

Feature operation is further broken down into three options:

1 Instant ISM parameter upgrade using a keycode diskette

2 Instant ISM parameter upgrade using HyperTerminal

3 Instant ISM parameter upgrade for Small Systems

Instant ISM parameter upgrade using a keycode diskette

Perform the following to instantly activate a keycode without a Sysload:

Note: For a dual-CPU (redundant) system, leave the system in full redundant mode (hard-disk and CPU redundancy).

1 Log in on a system terminal and access LD 143.

```
>LD 143
CCBR000
.
```

2 Insert the new keycode diskette into the floppy drive on the active core.

3 Enter the KDIF command and select keycode comparison options.

Note: Ensure that the new keycode does not lower ISM limits or reduce features compared with the existing keycode. If you have determined that the keycode lowers ISM limits or reduces features, do not continue with the KNEW command, but contact your Nortel Networks order management representative.

. KDIF
Please use: KDIF <param1> <param2>
with the following parameters:

NEW	accepted new keycode
REC	currently used keycode
OLD	previously used keycode
F0	candidate keycode on diskette in /f0 floppy drive
F1	candidate keycode on diskette in /f1 floppy drive
HD	candidate keycode which was uploaded to hard disk

Enter the keycode comparison option. The new keycode option is shown in **bold**.

Note: In the following example, the (REC) currently used keycode will be compared with the new keycode disk in floppy drive F0. The limits shown are for example purposes only.

.KDIF REC F0
Validating Keycode File /p/install/keycode.rec ... OK
Validating Keycode File /f0/keycode.kcd ... OK

System parameters	1st keycode:	2nd keycode:
System Serial Number	: 46XX	46XX
Software Version	: 2311	2311
System Type	: Option 61C	Option 61C
Call Processor	: CP68040	CP68040
Release	: 24	24
Issue	: XX	XX
NTI Order Number	:	
NT SDID - 1	:	
NT SDID - 2	:	
Date and Time of Manufacture	:	

Note: ( ) indicates that information is not available

ISM Limits	1st keycode:	2nd keycode:
Loop Limit	: 32	32
Sys TNs Limit	: 10	11
ACD Agt Limit	: 10	10
ACD DNs Limit	: 10	10
AST Limit	: 10	10

.....

Common packages for both keycodes:
0-2 4-5 7-25 28-29 32-55 58-65

.....

Additional packages in the 2nd keycode:
< **30-31**

.

4 Select the new keycode for activation using the KNEW command.

. KNEW F0

If the new keycode is eligible for instant activation, it will be activated without further user action, and the following system message is given:

CCBR020 New Keycode accepted and activated successfully. Sysload is NOT needed!

Otherwise, if the keycode is not eligible for instant activation, a Sysload is needed to activate the new keycode and the following system message is given:

CCBR009 New Keycode accepted. It will be activated during the next restart.

5 Access LD 22 and confirm that the new ISM parameters have been updated.

>LD 22
REQ SLT
....

6 See“Reverting to the previous keycode with the KRVR command” on page 616 if ISM limits are not increased or problems exist.

Instant ISM parameter upgrade using HyperTerminal®

For Large Systems, perform the following to instantly activate a keycode without a Sysload:

For a dual-CPU (redundant) system, leave the system in full redundant mode (hard-disk and CPU redundancy).

1 On a PC, access the system (using a modem) with HyperTerminal® (provided with Windows 95):

- Click the **Start button | Programs | Accessories | HyperTerminal.**

2 Double-click the HyperTerminal client to the system.

3 Log into the system.

4 Load the Keycode Management Program (LD 143).

LD 143	to load program
KUPL	to upload keycodes to the hard disk on the target system

5 Click the **Transfer** menu in HyperTerminal and select **Send Text File**.

6 From the **Files of type** pull-down menu, select **All Files (*.*)**.

7 Locate and select the keycode file on the PC. Use the **Look in** pull-down menu to select the drive on which the keycode is located.

8 Click **Open**.

The keycode will be displayed after the KUPL prompt.

Example:

KUPL 0001PBX 0101
9FPAMSRHNN17KRUQAFFSPREQEVMTHIDHRKDJHRKEJR56

9 Press the Enter key.

The Keycode is checked for CRC errors and is uploaded to the hard disk.

Enter the following command:

KDIF REC HD to compare the existing keycode with the new keycode on the hard disk

Ensure that the new keycode does not lower ISM limits or reduce features compared with the existing keycode. If you have determined that the keycode lowers ISM limits or reduces features, do not continue with the KNEW command, but contact your Nortel Networks order management representative.

10 Select the new keycode for activation using the KNEW command.

KNEW XX to select the new keycode for activation, where

XX = HD for a keycode on the hard drive, or
XX = F1 or F0 for a keycode on the floppy drive on Core 1 or Core 0.

If the new keycode is eligible for instant activation, it will be activated without further user action, and the following system message is given:

> CCBR020 New Keycode accepted and activated successfully. Sysload is NOT needed!

Otherwise, if the keycode is not eligible for instant activation, a Sysload is needed to activate the new keycode and the following system message is given:

> CCBR009 New Keycode accepted. It will be activated during the next restart.

If KUPL fails, the file is saved to the file "\u\keycode.err."

11 See "Reverting to the previous keycode with the KRVR command" on page 616 if ISM limits are not increased or problems exist.

Instant ISM parameter upgrade for Small Systems

For Small Systems, perform the following to instantly activate a keycode without a Sysload:

1 Log in and access LD 143

>LD 143
CCBR000
.

2 Enter the **UPGRADE** command.

. UPGRADE

The “Software Installation Main Menu” is displayed:

SOFTWARE INSTALLATION PROGRAM
****
Verify Security ID: XXXXXXXX
****
Software Installation Main Menu:
1. New Install or Option 11/11E Upgrade - from Software Daughterboard
2. System Upgrade
3. Utilities
4. New System Installation - From Software Delivery Card
[q]uit, [p]revious, [m]ain, [h]elp or [?], <cr> - redisplay

Enter Selection:

3 Enter **2** for the “System Upgrade” option.

The “Select type of upgrade to be performed” menu is displayed.

Select type of upgrade to be performed:
1. Option 11/11E to Option 11C
2. Option 11C New Software Upgrade
3. Option 11C Feature/Parameter Upgrade
[q]uit, [p]revious, [m]ain, [h]elp or [?], <cr> - redisplay

4 Enter **3** for the “Option 11C Feature/Parameter Upgrade” option.

Note: The following questions require information from the Keycode data sheet. Please have it available.

5 Indicate that the current Feature Sets and/or Packages will remain the same by selecting “n” to the following requests.

- Do you wish to change feature sets? (y/n/[a]bort) : N
  Keeping Current Feature Set.
- Do you wish to add packages? (y/n/[a]bort) : N

The current ISM Parameters are printed to the TTY.

6 The ISM parameters shown below are a sample configuration only.

Current ISM Parameters :
TNS (10)
AGNT (10)
ACDN (10)
AST (10)
DSL (10)
...

7 Do you wish to change any ISM parameters? (y/n/[a]bort) :

8 In response to the prompt “Do you wish to change any ISM parameters? (y/n/[a]bort) :” enter **y**.

9 The ISM parameters are prompted in sequence. Change the ISM parameters appropriately, according to the new keycode:

The ISM parameters shown below are a sample configuration only.

Enter new ISM parameters, <cr> to leave unchanged:
TNS (10) -
AGNT (10) - 11
ACDN (10) -
AST (10) -
DSL (10) -
...

10 After all ISM parameters have been prompted, the new ISM parameters are displayed and the prompt “Is this correct?” appears. Enter **y** to continue.

11 New ISM Parameters :
TNS (10) -
AGNT (11)
ACDN (10) -
AST (10) -
DSL (10) -
...

12 Is this correct? (y/n/[a]bort) :

13 Enter **y** if the new ISM parameters are correct. If the ISM parameters are not correct select **n** and reconfigure the ISM parameters.

The system will display the Security ID and Current AUX ID.

14 Security ID: XXXXXXXX
Current AUX ID : XXXXXXXX
Do you wish to change the AUX ID? (y/n/[a]bort) :

15 In response to the prompt “Do you wish to change the AUX ID?,” enter **n**.

16 An upgrade summary is displayed. In response to the prompt “Is this correct?,” enter **y** to continue.

17 Ensure that the new ISM limit is shown. In this example the AGNT ISM limit was changed from 10 -11. The system will display:
AGNT : 10 11
...
Is this correct? (y/n/[a]bort) :

18 Select **y** and the system will prompt to enter the keycode.

19 Enter new keycodes:
Key 1 :
Key 2 :
Key 3 :

20 Enter the new keycode. The keycode consists of three keycode strings: Key 1, Key 2, and Key 3. Enter each string and press return. If the keycodes are entered properly, the system will display:

```
Keycode validation successful.
Are you sure you wish to perform the upgrade? (y/n/[a]bort) :
```

21 In response to the prompt "Are you sure you wish to perform the upgrade?", enter **y**.

If the new keycodes correct for instant activation, it will be activated without further user action, and the following message is given:

```
Upgrade was completed and activated successfully.
Sysload is NOT needed!
```

If the keycode is not eligible for instant activation, a Sysload is needed to activate the new keycode and the following message is given:

```
Upgrade was completed successfully.
Initiate a Sysload to activate the upgrade.
```

Reverting to the previous keycode with the KRVR command

On Large Systems, the KRVR command can be used to revert to the old keycode "instantly."

Note: The terms "old" and "new" keycode as discussed here refer to the most recent previous KNEW command. The "old" keycode is the former keycode, prior to the KNEW command. The "new" keycode is the keycode that was activated by the KNEW command.

The old keycode is eligible for instant activation with the KRVR command if the only difference between the old keycode and the new keycode is that some or all of the ISM parameters in the old keycode are *higher*.

To revert to the old keycode:

- In LD 143, enter the **KRVR** command.

If the keycode is eligible for instant activation, it will be activated without further user action, and the following system message is given:

CCBR020 New Keycode accepted and activated successfully. Sysload is NOT needed!

Otherwise, if the keycode is not eligible for instant activation, a Sysload is needed to activate the new keycode and the following system message is given:

CCBR009 New Keycode accepted. It will be activated during the next restart.

Integrated Messaging System Link

Contents

This section contains information on the following topics:

Feature description 619
Operating parameters 620
Feature interactions 620
Feature packaging 621
Feature implementation 621
Feature operation 625

Feature description

The primary objectives of Integrated Messaging System (IMS) Link are to replace written telephone messages, to minimize the need for attendant intervention in the leaving and the retrieving of messages, and to support user-to-user automatic voice messaging. These functions are integrated in Integrated Messaging System (IMS) Link capability.

Integrated Messaging System (IMS) Link provides the support required for third-party messaging systems to interface with the system. The calling party can leave voice messages to be retrieved by the called party at any time. Users calling from inside or outside the system can leave and retrieve messages. The messaging system answers the call, delivers a personal greeting (recorded in the user's voice), digitizes the message, stores the message, and notifies the called party of a waiting message. The called party can retrieve and manipulate these messages from any Digitone telephone in the world. The user can issue a variety of commands to save or transfer messages, reply to messages, or broadcast group messages to multiple users.

To retrieve messages, each user must enter an ID code and a password. If the user calls the messaging system from his or her own Directory Number (DN), the ID code need not be entered. Any telephone with Dual Tone Multifrequency (DTMF) or Meridian 1 proprietary telephone signaling can connect to the attendant or to some other predefined DN by pressing 0. Callers with analog (500/2500 type) telephones must wait for a time-out before connecting automatically to the attendant.

The maximum length of a message will vary, depending on the messaging system equipped. User profiles are established to limit the number of messages each user is entitled to store.

Operating parameters

Users within the system must have either Dual Tone Multifrequency (DTMF), or Meridian 1 proprietary telephone signaling capabilities. Users outside the system must have DTMF signaling.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Integrated Messaging System (IMS) package 35, requires the following packages:

- Basic ACD (BACD) package 40
- ACD Package A (ACDA) package 45
- Message Center (MWC) package 46
- Auxiliary Processor Link (APL) package 109

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Add or change the link to a messaging system. Before adding, changing, or removing a link, the device must be disabled.

2 LD 17 – Add or change the link to a messaging system. Before adding, changing, or removing a link, the device must be disabled.

3 LD 15 – Add or change the IMS feature for a customer.

4 LD 23 – Add or change ACD data for Integrated Messaging System Link feature.

5 LD 11 – Add or change IMS attendant capability for each Meridian 1 proprietary telephone.

LD 17 – Add or change the link to a messaging system. Before adding, changing, or removing a link, the device must be disabled.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device And Number
IOTB	(NO) YES	(Do not) allow changes to input/output devices.
ADAN	NEW CHG TTY 0-15	Add or change a messaging system link to the system.
- USER	APL	This link is an Auxiliary Processor Link (APL).
TYPE	PARM	System Parameters
- AXQI	(20)-255	Number of call registers to be used for receipt of messages from the messaging system.
- AXQO	(20)-255	Number of call registers to be used for output of messages to the messaging system. ***Note:*** If the number of call registers defined for the system (prompt NCR) is within the range 80-1020, AXQI and AXQO cannot exceed 25 percent of the system call registers.

LD 17 – Add or change the link to a messaging system. Before adding, changing, or removing a link, the device must be disabled.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Action Device And Number
IOTB	(NO) YES	(Do not) allow changes to input/output devices.
ADAN	NEW CHG TTY 0-15	Add or change a messaging system link to the system.

- CTYP	aaaa	Card type, where: aaaa = DCHI, MSDL, MSPS, SDI, SDI2, SDI4, or XSDI.
- DNUM	0-15	Device number to be printed automatically (same as ADAN number).
- USER	APL	This link is an Auxiliary Processor Link (APL).
TYPE	PARM	System Parameters.
- AXQI	(20)-255	Number of call registers to be used for receipt of messages from the messaging system.
- AXQO	(20)-255	Number of call registers to be used for output of messages to the messaging system. ***Note:*** If the number of call registers defined for the system (prompt NCR) is within the range 80-1020, AXQI and AXQO cannot exceed 25 percent of the system call registers.

LD 15 – Add or change the IMS feature for a customer.

Prompt	**Response**	**Description**
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
OPT	(MCX) MCI	Message Center (excluded) included.
...		
TYPE	IMS	Integrated message service options.
- IMS	(NO) YES	(Do not) allow changes to the IMS feature.
- IMA	(NO) YES	IMS feature (is not) or is enabled.
- - APL	0-15	Port number of the link to the messaging system.

- UST	(NO) YES	User Status Update (UST) feature (is not) or is enabled.
- - APL	0-15	Port number of the link from UST to the messaging system.
- UMG	(NO) YES	User-to-User Messaging (UMG) feature (is not) enabled.
- - APL	0-15	Port number of the link from UMG to the messaging system.

LD 23 – Add or change ACD data for Integrated Messaging System Link feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ACD	ACD Data Block.
CUST	xx	Customer number, as defined in LD 15
ACDN	xxxx	ACD DN (can have up to seven digits if DN Expansion package is equipped).
MWC	(NO) YES	ACD (is not) is an IMS.
- IMS	(NO) YES	(Do not) allow changes to the IMS feature.
- - IMA	(NO) YES	ACD DN (is not) is used as an IMS DN.
- - APL	0-15	Port number of the link to the messaging system.
- - UST	(NO) YES	User Status Update (UST) feature (is not) is enabled.
- - APL	0-15	Port number of the link from UST to the messaging system.
- - UMG	(NO), YES	User-to-User Messaging (UMG) feature (is not) is enabled.
- - APL	0-15	Port number of the link from UMG to the messaging system.
- - RAN	0-30 32-xxx	Route number to the Recorded Announcement (RAN) for UMG (default is no RAN).
- - UMT	0-(6)-15	Time, in seconds, of silent interval after alert tone on RAN.

LD 11 – Add or change IMS attendant capability for each Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:		Telephone type, where:
	aaaa	aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(IMD) IMA	This telephone (is not) is an IMS attendant.
LTN	1-253 0-15	Logical Terminal Number assigned to this attendant, port number of the link to messaging system used by this attendant.
KEY	0 ACD xxxx yyyy	Add an INCALLS key, where: xxxx = IMS Directory Number (DN), and yyyy = Agent ID. ***Note:*** IMS DN and Agent ID can have up to seven digits if DN Expansion package is equipped
	xx MIK xx MCK xx NRD xx MSB	Add a Message Indication (MI) key. Add a Message Cancellation (MC) key. Add a Not Ready (NR) key. Add a Make Set Busy (MSB) key.

Feature operation

No specific operating procedures are required to use this feature.

Integrated Services Digital Network

Integrated Services Digital Network (ISDN) provides standard digital interfaces between telephones, terminals, and telecommunication networks.

ISDN uses a common signaling protocol transmitted over a dedicated data channel called the D-channel. The D-channel carries call setup and feature activation information to the call destination. This allows users network-wide access to features.

ISDN services are categorized into two types of interfaces: Primary Rate Interface (PRI) and Basic Rate Interface (BRI).

Primary Rate Interface (PRI)

ISDN PRI provides 30B+D 23B+D channels, offering digital connectivity between the system and supported interfaces.

For more information on ISDN PRI, please refer to the ISDN Primary Rate Interface NTPs.

Basic Rate Interface (BRI)

ISDN BRI is a digital connection that provides three digital channels. These channels consist of two 64 kbps bearer channels (B-channels) and one 16 kbps signaling channel (D-channel). This 2B+D connection is known as a Digital Subscriber Link (DSL). The DSL can be configured to provide line access, trunk access, or packet data transmission.

For more information on ISDN BRI, please refer to the ISDN Basic Rate Interface NTPs.

Integrated Voice and Data

Contents

This section contains information on the following topics:

Feature description 629
Operating parameters 629
Feature interactions 630
Feature packaging 630
Feature implementation 630
Feature operation 635

Feature description

The Integrated Voice and Data feature provides integrated voice and data switching through a host system.

Hardware consists of the Add-on Data Module (ADM), Data Line Card (DLC), and Modem Pool Line Card (MPLC), if modem pooling is used.

The system software recognizes the ADM as an SL-1 telephone, the DLC as an SL-1 Line Card, and the MPLC as a 500 telephone Line Card. LD 10 and LD 11 are used to enter the hardware into the office data.

Operating parameters

Hunting is not allowed with the Modem Pool Line Card (MPLC) pack.

No analog (500/2500 type) telephone can be assigned to the MPLC pack.

Collocated SL-1 telephones can only have three key/lamp strips, due to physical constraints.

Feature interactions

For more information on Integrated Voice and Data, refer to *Meridian Data Features: Operations and Tests* (553-2731-300).

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Add or change the SL-1 telephone (of an SL-1 telephone/Add-on Data Module pair) associated with a Data Line Card (DLC) data port pair.

2 LD 11 – Add or change ADM (of an SL-1 telephone/ADM pair) associated with a Data Line Card (DLC) data port pair.

3 LD 11 – Add or change the DLC data port associated with a standalone ADM.

4 LD 11 – Add or change the Integrated Data Interface Card (IDLC) port associated with an Asynchronous Interface Module (AIM).

5 LD 16 – Define the trunk route for each data port group (modem pool).

6 LD 14 – Define a DLC as a trunk for each data port within the data port group.

7 LD 10 – Define a Modem Pool Line Card (MPLC) for each modem in the data port group.

8 LD 16 – Define a route data block for each Central Office (CO), FEX, TIE, or WATS trunk route to a remote system.

9 LD 14 – Define each trunk within the route.

LD 11 – Add or change the SL-1 telephone (of an SL-1 telephone/Add-on Data Module pair) associated with a Data Line Card (DLC) data port pair.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	SL1	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems SL-1 telephones are restricted to unit 0 or 2 when collocated with an ADM.
CDEN	SD DD	Density of this card is single or double.
KLS	1-7	Number of key/lamp strips.
KEY	0 DN xxx...x 2 TRN 9 RLS	Key 0; Voice Frequency Directory Number. Key 2; Call Transfer key. Key 9; Release key. ***Note:*** Other feature keys may be associated as required, subject to the limitations imposed by the companion ADM.

LD 11 – Add or change ADM (of an SL-1 telephone/ADM pair) associated with a Data Line Card (DLC) data port pair.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	SL1	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems The loop, shelf, and card must be the same as those specified for the companion SL-1 telephone. The unit must be the next subsequent unit to the companion SL-1 telephone (for example, if the unit for SL-1 telephone is 2, the unit for ADM must be 3).

CDEN	SD DD	Single or double density card.
CLS	WTD	Warning Tone Denied.
KEY	0 DN xxxx	Key 0, data Directory Number; can have up to seven digits if DN Expansion (DNXP) package is equipped.
	1 DN xxx...x	Key 1, optional secondary data DN.
	2 TRN	Key 2, Call Transfer key (optional).
	3 ADL x...x	Key 3, Autodial DN (optional).
	4 RGA	Key 4, Ring Again key (optional).
	6 SCC 0-8190 *or* 6 SCU 0-8190	Speed Call Controller, Speed Call List number (optional; must be on key 6 if equipped). Speed Call User, Speed Call List number (optional; must be on key 6 if equipped).
	9 RLS	Release key: must be key 9. ***Note:*** Only the feature keys listed above can be assigned to the Add-on Data Module (ADM). If they are assigned to the ADM, they must also be assigned to the companion SL-1 telephone on the same keys(i.e, if the ADM has ADL on key 3, the companion SL-1 telephone must also have ADL on key 3, with the same Autodial DN).

LD 11 – Add or change the DLC data port associated with a standalone ADM.

Prompt	**Response**	**Description**
REQ:	CHG	Change.
TYPE:	SL1	Telephone type.
TN		Terminal Number
	l s c u	For Large Systems
	c u	For Small Systems and Succession 1000 systems
CDEN	SD	Single density card.
CLS	WTD	Warning Tone Denied.
KEY	0 DN xxx...x	Key 0, data Directory Number.
	9 RLS	Key 9, Release key.
		Note: Other features/functions must not be assigned to keys 1-8.

LD 11 – Add or change the Integrated Data Interface Card (IDLC) port associated with an Asynchronous Interface Module (AIM).

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	SL1	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems For AIM, unit 1 or 3 should be used.
CDEN	SD	Single density card.
CLS	WTD	Warning Tone Denied.
KEY	0 DN xxx...x 1 DN xxx...x 2 TRN 3 ADL x...x 4 RGA 6 SCC 0-8190 *or* 6 SCU 0-8190 9 RLS	Key 0, data Directory Number. Key 1, optional secondary data DN. Key 2, Call Transfer key (optional). Key 3, Autodial DN (optional). Key 4, Ring Again key (optional). Speed Call Controller, Speed Call List number (optional; must be on key 6 if equipped). Speed Call User, Speed Call List number (optional; must be on key 6 if equipped). Release key; must be key 9.

LD 16 – Define the trunk route for each data port group (modem pool).

Prompt	Response	Description
REQ	NEW CHG	Create a new route, or modify an existing one.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15

ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	ADM	ADM route.
ACOD	xxx...x	Access code for this route.
CDPC	(NO) YES	SL-1 (is not) is the only controlling party on incoming calls.

LD 14 – Define a DLC as a trunk for each data port within the data port group.

Prompt	Response	Description
REQ	NEW CHG	Create a new trunk or modify an existing one.
TYPE	ADM	ADM trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 10 – Define a Modem Pool Line Card (MPLC) for each modem in the data port group.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD DD 4D	Single, double, or quad density card.
DN	xxx...x	Voice Frequency Directory Number; must be the same as that telephone by switches in the ADM. ***Note:*** The trunk route defined for the data port group in LD 16 cannot be used.

LD 16 – Define a route data block for each Central Office (CO), FEX, TIE, or WATS trunk route to a remote system.

Prompt	Response	Description
REQ	NEW, CHG	Create a new route, or modify an existing one.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	COT FEX TIE WAT	Route type.
ACOD	xxx...x	Access code for the route.

LD 14 – Define each trunk within the route.

Prompt	Response	Description
REQ	NEW CHG	Create a new trunk or modify an existing one.
TYPE	COT FEX TIE WAT	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD DD	Single or double density card.

Feature operation

No specific operating procedures are required to use this feature.

Intelligent Peripheral Equipment Completion

Contents

This section contains information on the following topics:

Feature description 637
Operating parameters 638
Feature interactions 638
Feature packaging 639
Feature implementation 639
Feature operation 645

Feature description

RON/TRON Signaling on XFEM

RON/TRON signaling is required for the Italian Extended Flexible E&M card (XFEM). RON/TRON is similar in operation to the current E&M signaling, the difference being that instead of an Answer Acknowledge, a Seize Acknowledge is sent by the far end and it remains for the duration of the call.

L1 Signaling on XFEM

L1 is a signaling protocol for inter-circuit switched network connections defined by the International Telegraph and Telephone Consultative Committee (CCITT) Q8 recommendation. This signaling is similar to AC15, but introduces two new signals: Seize Acknowledge; and Proceed to Send.

LDR Signaling on Italian DID card (XIDID)

It will be possible to configure Loop Dial Repeat (LDR) signaling on a TIE trunk on an XDID card. LDR signaling on a TIE trunk with an XIDID card operation is similar to LDR signaling on a TIE trunk.

Operating parameters

The following hardware cards are required:

- XFEM – NT5K83GA (for RON/TRON), NT5K83HB (for L1 in Belgium), or NT5K83DB (for L1 in Holland)
- XIDID – NTCK22AA

Feature interactions

B34 Codec Static Loss Download and B34 Dynamic Loss Switching

Whenever a TIE/LDR trunk is configured on an XIDID card, for Static Loss Plan Download (SLPD)/Dynamic Loss Switching (DLS), loss/level is downloaded/switched to an XDID card with the type 12 message. Depending on the Class of Service configured, Non-Transmission Compensated (NTC), Transmission Compensated (TRC), or Via Net Loss (VNL), the TIE unit will be mapped to the following B34 port types: B34 T2WN, B34 T2WT, or B34 T2WV.

Multifrequency Compelled Signaling (MFC)
Multifrequency Compelled Signaling for Socotel (MFE)

MFC, MFE, L1 signaling and RON/TRON signaling are mutually exclusive.

Tone to Last Party

This feature provides a special tone (default value is busy tone) to both analog (500/2500 type) telephones and trunks in half disconnect state. The operation of this feature is unchanged for trunks working with L1 or RON/TRON.

Trunk-to-Trunk Connection

The existing restrictions for trunk-to-trunk connections based on trunk type will be applicable to XFEM cards using L1 or RON/TRON signaling.

Partial Dial Timer

This feature limits the interdigit delay to the value of the End-of-dial (EOD) timer, and its functionality is extended to TIE trunks with L1 or RON/TRON signaling.

Feature packaging

There are no new packages introduced with this feature; however, RON/TRON, L1, and LDR on XDID will be packaged with Intelligent Peripheral Equipment (XPE) package 203.

Feature implementation

RON/TRON Signaling on XFEM

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Configure a RON/TRON signaling trunk route.

2 LD 14 – Configure RON/TRON Signaling trunk.

LD 16 – Configure a RON/TRON signaling trunk route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
...		
TKTP	TIE	TIE trunk.
CNTL	YES	Change control or timers.
- TIMR		Timer.
	DDL 0-(70)-1023	Dial Delay timer. The DDL timer is set at 512 ms. for the RT (RON/TRON) start arrangement.
	DSI 128-(34944)-499200	Disconnect Supervision timer.
	EOD 128-(13952)-32640	End-of-dial timer.
	ICF 0-(512)-32640	Incoming Flash timer.
	OGF 0-(512)-32640	Outgoing Flash timer. The OGF timer is to be set to 384 ms. for validation of the seize acknowledge message.
	SST xx	Seizure Supervision timer for trunks with delay dial (DDL), wink (WNK), and ground (GRD) start arrangements. xx = a minimum value of 1-(3)-15 seconds for GRD, and five seconds for DDL, WNK, RT (RON/TRON) start arrangement, and L1 signaling.
DTD	YES	Dial tone detection.
MDTD	1-(5)-31	Minimum dial tone detection delay for the route in seconds.
DLTN	(NO) YES	Provide dial tone to the far end.

LD 14 – Configure RON/TRON Signaling trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data.
TYPE	TIE	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XFEM	Extended Flexible E&M trunk card.
SIGL	EAM	E&M two-wire.
...		
STRI	RT	RON/TRON incoming signaling start arrangement.
STRO	RT	RON/TRON outgoing signaling start arrangement.
CLS	DTN	Digitone.

L1 Signaling on XFEM

Task summary list

The following is a summary of the tasks in this section:

1. LD 16 – Configure L1 signaling on a XFEM TIE trunk route with Proceed to Send expected after an outgoing seize and answer supervision.
2. LD 14 – Configure a L1 Signaling trunk.

LD 16 – Configure L1 signaling on a XFEM TIE trunk route with Proceed to Send expected after an outgoing seize and answer supervision.

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data.
TYPE	RDB	Route data block.
...		
TKTP	TIE	Trunk type.
CNTL	YES	Change control or timers.
- TIMR		
	DDL 0-(70)-1023	Dial Delay timer.
	DSI 128-(34944)-499200	Disconnect Supervision timer.
	EOD 128-(13952)-32640	End-of-dial timer.
	ICF 0	Incoming Flash timer.
	OFC 0	Outgoing Flash timer.

SST	xx	Seizure Supervision timer for trunks with delay dial (DDL), wink (WNK), and ground (GRD) start arrangements. xx = a minimum value of 1-(3)-15 seconds for GRD, and five seconds for DDL, WNK, RT (RON/TRON) start arrangement, and L1 signaling.
DTD	NO	Dial Tone Detection.
MDTD	1-(5)-31	Minimum Dial Tone Detection Delay for route in seconds.
DLTN	NO	Provide Dial Tone to the far end.

LD 14 – Configure a L1 Signaling trunk.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data or change existing data.
TYPE	TIE	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XFEM	Extended Flexible E&M Trunk Card.
SIGL	WR4	AC15 Four-wire signaling; CEPTL1 Signaling.
...		
STRI	PTSD	Proceed-to-send to be sent upon receipt of an incoming seize.
STRO	PTSD	Proceed-to-send expected after generation of an outgoing seize.
SUPN	YES	Answer Supervision.
CLS	DTN	Digitone.

LDR signaling on XIDID

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Configure LDR signaling on XIDID trunk route.

2 LD 14 – Configure LDR signaling on XIDID trunk.

LD 16 – Configure LDR signaling on XIDID trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data.
TYPE	RDB	Route data block.
...		
TKTP	TIE	Trunk type.
...		

LD 14 – Configure LDR signaling on XIDID trunk.

Prompt	Response	Description
REQ	NEW	New.
TYPE	TIE	TIE trunk data block.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XDID	Extended DID trunk card.
SIGL	LDR	Loop Dial repeating.
LDOP	LOOP	Loop outpulsing for LDR signaling.
BIMP	600	Balance impedance 600 ohms.

STRI	IMM	Immediate incoming start arrangement.
STRO	IMM	Immediate outgoing start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	NTC	Non-transmission compensated.

Feature operation

No specific operating procedures are required to use this feature.

Intelligent Peripheral Equipment Software Support Enhancements

Contents

This section contains information on the following topics:

Feature description 647
Operating parameters 648
Feature interactions 649
Feature packaging 649
Feature implementation 650
Feature operation 652

Feature description

This feature provides software enhancements to the XFEM, XFALC, XFCOT, XDID, and XTD cards. The new functionalities are as follows:

- XFEM – An E&M signaling type is introduced for EAM or EM4/WR4 configurations. The BPO signaling type can be selected as an answer to the EMTY prompt. BPO is sometimes referred to as Type V signaling.
- XFALC – Some previously hard-coded timers can now be configured on a per-system basis, including off-hook validation, minimum time for dial pulse, interdigit timer, maximum time for dial pulse, and the existing post-flash timer.

- XFCOT – The Autoguard function is enhanced with an Autoguard Repeat Prevention (ATP) timer. This timer denies outgoing calls on a trunk after seize failure during the time configured for ATP. Fastguard functionality is added to prevent call collision between incoming and outgoing calls. If a Fastguard message is received from a Central Office Trunk (COT), the trunk unit is made busy immediately, thus avoiding any outgoing call to seize this unit which would drive it back to a glare state.
- XDID – This allows the Balance Impedance Adjustment to be configurable and downloadable.
- XTD – Auto configuration of the XTD card with the XTD Table 0 parameters can now be enabled. If different parameters are required for a specific XTD card, a new XTD table must be configured in LD 97. This specific card has to be manually reconfigured with the newly defined XTD table.

Operating parameters

The BPO signaling type is downloaded to the XFEM card using two hardware IDs: EAM_BPO and EM4_BPO. These IDs are supported on the Dutch XFEM card NT5K83DA, and the Italian XFEM card NT5K83GA.

The flexible XFALC Timer Download is supported on the country-specific XFALC cards NT5K20XX, where XX is the country-specific suffix.

Fast guard is supported on the New Zealand NT5K18BA, and Australia NT5K82BA/CA XFCOT cards.

The ARP timer for enhanced Autoguard applies only to Intelligent Peripheral Equipment (IPE) analog loop-start CO trunks.

XTD auto configuration is supported on the global XTD card NT5K48AA.

The Fastguard functionality only applies for incoming Loop Start CO trunks.

Auto configuration of an XTD card takes place:

- If an XTD card is inserted in a slot of an IPE shelf for which nothing is configured in the software. In such a case, XTD Table 0 parameters are used, and all units have DTD and DTR capability.
- If an XTD card is inserted in a slot of an IPE shelf for which at least one XTD unit is already configured in software. In that case, all non-defined units are automatically configured with the same XTD Table number as the unit(s) that are already defined. The newly configured units have DTD and DTR capability.

Feature interactions

The XFCOT has the following interactions with Loop Start Public Exchange/ Central Office trunks:

- Fastguard – seizure of an incoming trunk can be done by sending either a Ring Burst or Fastguard message from the firmware to the software.
- ARP – the ARP timer replaces the hard coded 3s timer.
- The Office Data Administration System (ODAS) provides a method of retrieving administrative information stored in system memory, such as the date that a feature package was last modified by a service change. Pertaining to XTD, whenever an XTD unit is created with Auto configuration, the system date when Auto configuration took place is stored at the end of the terminal number (TN) list.

Feature packaging

Intelligent Peripheral Equipment Software Support Enhancements require Intelligent Peripheral Equipment (XPE) package 203. The following packages are also required:

- Multi-party Operations (MPO) package 141
- International Supplementary Features (SUPP) package 131
- Automatic Card Installation (AINS) package 200
- Dial Tone Detector (DTD) package 138

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Configure XFEM BPO trunk type signaling, and the Balance Impedance Adjustment on XDID trunk.

2 LD 16 – Configure the Autoguard Repeat Prevention timer for the route.

3 LD 97 – Configure the five XFALC timers to support downloading.

LD 14 – Configure XFEM BPO trunk type signaling, and the Balance Impedance Adjustment on XDID trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
XTRK	XFEM	Extended E&M trunk card.
...		
SIGL	EAM EM4 WR4	E&M 2, 4 wire and AC15 4 wire.
...		
EMTY	(ty2) ty1 BPO XBPO	4 wire E&M (type 2) or type 1 or BPO. XBPO is used to suppress the BPO trunk type and signaling option in case of EM or WR4 type signaling.
XTRK	XDID	Extended DID trunk card.
...		
BIMP	(3COM) 600	Balance impedance.

LD 16 – Configure the Autoguard Repeat Prevention timer for the route.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
TYPE	RDB	Route Data Block.
...		
CNTL	YES	Responding YES to this prompt will display the TIMR prompt below.
TIMR	ARP 1-(3)-255	Autoguard Repeat Prevention timer. For Australia, the recommended value of ARP is 200 seconds.

LD 97 – Configure the five XFALC timers to support downloading.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
FLSH	xxx yyyy	Flash timing
TOHV	0-(250)-1275	Off-hook validation timer, in milliseconds.
TDP	(15)-1275	Minimum time for dial pulse, in milliseconds.
TID	0-(150)-1275	Interdigit timer, in milliseconds.
TDPO	15-(150)-1275	Maximum time for dial pulse, in milliseconds.
TPF	0-(200)-1275	Post-flash timer, in milliseconds. Prompted only if MPO is equipped.

Note: For Timer Settings, the value set for the TDP timer must be less than or equal to the setting for the switchhook flash timer. The TDPO timer must be greater than the TDP timer. All timer values must be entered in five milliseconds increments. Otherwise, the value is rounded to the closest inferior multiple of five.

Feature operation

No specific operating procedures are required to use this feature.

Intercept Computer Interface

Contents

This section contains information on the following topics:

Feature description 653
Operating parameters 655
Feature interactions 655
Feature packaging 656
Feature implementation 656
Feature operation 660

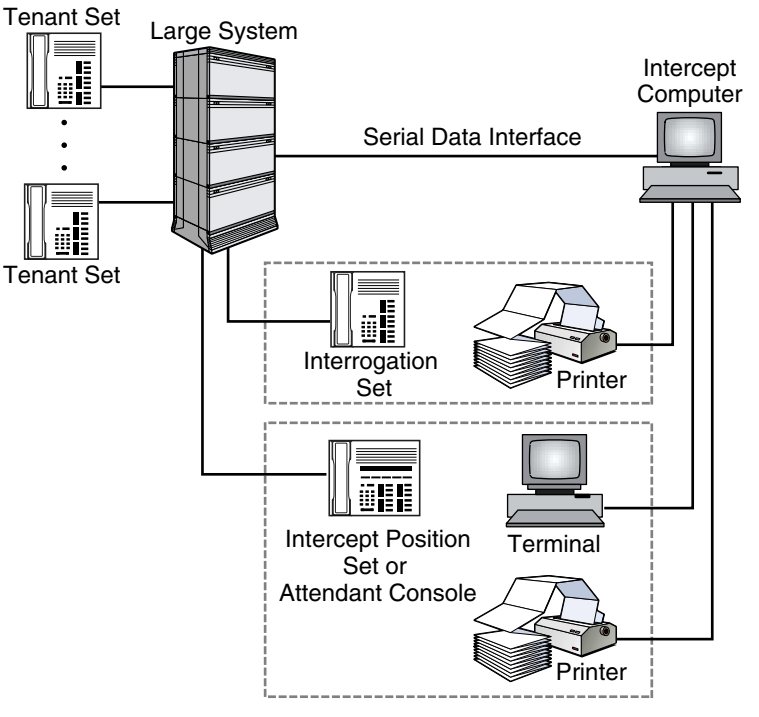
Feature description

This feature allows the system to use an intercept (attendant assistance service) computer for storing and retrieving call messages. Calls to an absent tenant's directory number (DN) using this feature are routed to a designated Intercept Position (ICP) DN.

The feature can be activated or deactivated by the following:

- A Flexible Feature Code (FFC) dialed from the tenant's telephone. This code specifies the reason for the tenant's absence and can be extended with a date and time as extra information. The FFC decodes into a text message.
- Pressing the Call Forward All Calls (CFW AC) key on an SL-1 telephone (deactivation).

Figure 16
Intercept Computer Interface components



- From the ICP terminal.
- Automatically when a terminal number (TN) is disabled or enabled by a maintenance overlay program.

The feature is available to all analog (500/2500 type) telephones and Meridian 1 proprietary telephones. Any analog (500/2500 type) telephone can be designated to be an interrogation set. This is given a special FFC to allow the printing of messages for any or all DNs. The attendant (ATT), and Meridian 1 proprietary telephones can be used as an ICP.

A multiple channel answering machine can be connected to both the system and Intercept Computer. The machine is defined in the system as a Group Hunt list, and the Pilot Directory Number (PLDN) is used to terminate on the Answering Machine after a call has been diverted by the ICP feature. A 2500-type set may be designated as a channel in the ICP answering machine in LD 10. The set must have a Digitone (DTN) Class of Service.

Operating parameters

An analog (500/2500 type) telephone can only be used as an interrogation set, not as an ICP.

The number of ports available to the intercept computer is typically less than 12 (the number of TTY ports less those used for maintenance, service change, and traffic).

The CFW AC LED on the tenant's telephone is used to indicate both the CFW AC and this feature.

It is not possible to change or remove an ICP station by way of the LD 71 and 72.

Each SL-1 telephone must have one CFW AC key and possibly one message-waiting key on the LED key lamp strip (if the tenant requires this type of indication). These two LEDs are turned off automatically when the Intercept Computer Interface feature is deactivated (by dialing a FFC).

This feature makes use of the Message Center (MC) and Automatic Call Distribution (ACD) features. The ICP must be configured as an MC ACD DN or MC attendant DN.

This feature and CFW AC feature are not to be activated at the same time.

ICP and Integrated Messaging Services (IMS) cannot be used at the same time for the same customer.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the ICP Call Forward feature. If the dialed DN of the set that has the ICP Call Forward feature active is idle, the DN will be blocked and if the DN is busy, busy tone will be heard.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Intercept Computer Interface feature is not supported in a DPNSS1 UDP network.

Feature packaging

Intercept Computer Interface (ICP) package 143.

Dependencies:

- Automatic Call Distribution Package A (ACDA) package 45
- Auxiliary Processor Link (APL) package 109
- Flexible Feature Codes (FFC) package 139
- Flexible Tone and Cadences (FTC) package 125
- Message Waiting Center (MWC) package 46, and
- International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure the configuration record for Intercept Computer Interface.

2 LD 15 – Configure the customer data block for Intercept Computer Interface.

3 LD 10 – Create or modify the analog (500/2500 type) telephone data block for Intercept Computer Interface.

4 LD 11 – Create or modify the Meridian 1 proprietary telephone data block for Intercept Computer Interface.

5 LD 12 – Create or modify the Attendant Console data block for Intercept Computer Interface.

6 LD 23 – Modify the ACD/Message Center parameters for Incoming Call Indicators (ICIs).

7 LD 93 – Enable or modify the Multi-tenant Service feature.

LD 17 – Configure the configuration record for Intercept Computer Interface.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration data block.
IOTB	(NO) YES	Change to logical units.
ADAN	NEW TTY x	Add TTY number x.

LD 15 – Configure the customer data block for Intercept Computer Interface.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	CDB	Customer Data Block.
...		
- OPT	MCI	Message center included.
TYPE	IMS	Gate opener.
...		
- IMS	YES	Integrated messaging services excluded.
TYPE	ICP	Gate opener.
...		
- ICP	(NO) YES	Intercept Computer is (is not) available.
- NIPN	0-99	Number of intercept positions.

LD 10 – Create or modify the analog (500/2500 type) telephone data block for Intercept Computer Interface.

Prompt	Response	Description
REQ:	NEW, CHG	Add, or change.
TYPE:	500	500/2500Telephone data block.
...		
CLS	(IRGD) IRGA	Interrogation set for Intercept Computer allowed (denied).
	(IAMD) IAMA	Allow a 2500-type set to be a channel in the ICP Answering Machine (CLS DTN is required).
ICT	0-NIPN	Terminal/printer number (NIPN configured in LD 15).

LD 11 – Create or modify the Meridian 1 proprietary telephone data block for Intercept Computer Interface.

Prompt	Response	Description
REQ:	NEW, CHG	Add, or change.
TYPE:	a...a	Type of data block.
...		
CLS	(IPND) IPNA	Terminal/printer number (NIPN configured in LD 15).
ICT	0-NIPN	Terminal/printer number (NIPN configured in LD 15).

LD 12 – Create or modify the Attendant Console data block for Intercept Computer Interface.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	a...a	Attendant data block.
...		

ICP	(NO) YES	Intercept Computer (is not) is available.
ICT	0-NIPN	Terminal/printer number (NIPN configured in LD 15).

LD 23 – Modify the ACD/Message Center parameters for Incoming Call Indicators (ICIs).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ACD	ACD data block.
...		
ICP	(NO) YES	ACD MC (is not) is an intercept position.
ICPS		Intercept Computer printer search.
	COM	Common printer for ACD group.
	(CIR)	Circular hunt.

LD 93 – Enable or modify the Multi-tenant Service feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	a...a	Type of data block.
...		
ICP	(NO) YES	ACD MC (is not) is an intercept position.
ICPS		Intercept Computer printer search (when more than one console is used).
	(CIR)	Circular search.
	COM	One common printer for all consoles.

Feature operation

A terminal at the ICP displays a message stating why the tenant at the DN is absent. The person at the ICP can then store the caller's message for the tenant's DN and activate the message waiting LED at the tenant's telephone. The tenant at the DN retrieves the stored caller messages by calling the ICP, where the messages are displayed on the terminal (or optionally printed).

Intercept Computer Dial from Directory

Contents

This section contains information on the following topics:

Feature description 661
Operating parameters 662
Feature interactions 662
Feature packaging 667
Feature implementation 668
Feature operation 669

Feature description

An Intercept Computer (ICP) is an external information system that can be added to enhance attendant operation. Whenever an attendant answers an internal direct call, or any redirected call due to ICP Call Forwarding (CFW), the Intercept Computer Terminal (ICT) screen is lit up with information regarding either the caller (for internal calls), or the “called” party (for redirected calls). This information is presented to the attendant who can then give the appropriate information to the caller. For an external call, no information is displayed on the ICT screen.

With the Intercept Computer Dial from Directory feature (ICPD), the attendant does not need to dial the DN from the Attendant Console; pressing a single key on the ICT keyboard connects the call to the DN, thereby saving the attendant time.

An ICP can be programmed with a directory of the internal Directory Numbers (DNs) in the system. From the ICT, the attendant can search the ICP database for a specific person by name, in order to find that person's DN, according to a coordinated dialing plan (CDP). Again, pressing a single key on the ICT keyboard connects the call to the corresponding DN.

This feature is implemented using LD 15.

Operating parameters

The ICP feature must be configured for all related customers, and the ICP computer must be configured with the DNs that exist for these customers.

This feature is not available for either of the following intercept positions: ACD Agents; or the ICP Answering Machine.

It is only possible to dial from the ICT if the active loop is idle, or has only one part established in a call with the attendant (on Source (SRC) or Destination (DEST) side).

This feature does not support any dialing plan, other than CDP (since this is an already existing limitation of the networking part of the ICP feature).

A maximum of seven digits per incoming message can be received by the circuit switched network.

Feature interactions

Pre-dial Operations

Attendant Barge-in

It is possible for an attendant to Barge-in, in the following manner:

- Press an idle loop key, and press the Barge-in key from the Attendant Console.
- Dial a Route Access Code and Route Member from the ICT.

Attendant Busy Verify

It is possible for an attendant to Busy Verify in the following manner:

- Press an idle loop key, and press the Busy Verify key on the Attendant Console, and
- Dial an extension DN from the ICT.

Pre-dial Break-in

It is possible for an attendant to override call forward on a set in the following manner:

- Press an idle loop key, and press the Break-in key on the Attendant Console.
- Dial an extension DN from the ICT.

Call Forward/Hunt Override via Flexible Feature Code

Call Forward Hunt Override via Flexible Feature Code can be dialed prior to dialing the DN from the ICP.

Call Park

An attendant can park a call in the following manner:

- Press the Call Park key on the Attendant Console.
- Dial a DN from the ICT.
- Terminate Call Park operation by pressing the Release key.

Radio Paging Pre-dial Selection

It is possible to start automatic paging in the following manner:

- Dial the pre-dial selection RPA FFC on the Attendant Console.
- Dial a DN from the ICT.

Manual radio paging is started as follows:

- Dial the pre-dial selection RPA FCC on the Attendant Console.
- Dial a DN from the ICT.
- Dial a mode digit, digit information and octothorpe "#" sign.

Post-dial Operation

Attendant Break-in

An attendant can break-in to a call by:

- Dialing an extension DN from the ICT.
- Pressing the Break-in key on the Attendant Console.

Automatic Wake-up

This feature can be requested as follows:

- Press the Wake-up key on the Attendant Console.
- Dial a DN from the ICT.
- Dial an octothorpe sign "#", and terminate by dialing the requested wake-up time from the Attendant Console.

The same approach is used to cancel Automatic Wake-up.

Radio Paging Post-dial Selection

To start radio paging an extension DN:

- Dial a DN from the ICT.
- Press the RPA Post-dialing Paging (RPAG) key on the Attendant Console.

Stored Number Redial

An attendant can dial an extension from the ICT, and then press the Stored Number Redial key to store the called number (following the rules of the Stored Number Redial feature).

Other Feature Interactions

Attendant Recall with Splitting

If a set transfers a call to the attendant, or a Meridian 1 proprietary telephone presses the Attendant Recall (ARC) key and the transferring party has not yet completed the transfer before the attendant answers, it is not possible to dial from the ICP (since the transferred party is connected to SRC, and the transferring party is connected to DEST).

Autodial

It is possible to press the Autodial (ADL) key (in which some digits are stored such as an Electronic Switched Network (ESN) code or Flexible Feature Code (FCC)), and then dial a DN from the ICP. The DN will then be stored on the ADL key.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion can be activated by dialing an extension DN from the Intercept Computer Terminal, and then pressing the BKI key on the Attendant Console.

Do Not Disturb

This feature can be activated for an extension DN as follows:

- Press an idle Loop key, and press the Do Not Disturb Individual (DND IND) key on the Attendant Console.
- Dial a DN from the ICT.
- Press the DND IND key once more, and terminate the procedure by pressing the Release key on the Attendant Console.

The same approach applies when cancelling Do Not Disturb for a set.

To override Do Not Disturb for an extension DN:

- Press an idle Loop key on the Attendant Console.
- Dial a DN from the ICT.
- Press the DND IND key on the Attendant Console.

Message Waiting Indication

To activate the message waiting lamp:

- Press the Loop key and the Message Indication (MSG INDIC) key on the Attendant Console.
- Dial the set's DN from the ICT.
- Press the Message Indication key and the Release key on the Attendant Console.

The same approach can be used to turn off a Message Waiting lamp by using the Message Cancel key instead of the MSG INDIC key.

Multi-Tenant Service

The ICP Dial from Directory feature only works at the customer level. If several tenants are configured for a customer, they will all be affected by the ICTD prompt in LD 15.

Network Tenant Service

The ICP Dial from Directory feature only works at the customer level and for a single node. If several tenants are configured in a network situation, they will all be affected by how the ICTD prompt in LD 15 has been configured for the customers on different nodes.

Night Key Position Busy

If the Attendant Console has the Night key activated (for instance, it is busy or in Night Service), it is still possible to dial from the ICT.

Slow Answer Recall Enhancement

If the attendant extends an SRC party to a DEST party on the local node, but slow answer recall occurs since the DEST does not answer, it is possible to dial a new DN from the ICP (the DEST is disconnected when the attendant answers).

Transfer to Attendant

If a set transfers a call to the attendant, and the transferring party has not yet completed the transfer before the attendant has answered, dialing from the ICP is ignored (the transferred party is connected to SRC, and the transferring party is connected to DEST due to the Attendant Recall with Splitting feature).

Feature packaging

This feature is packaged under the Intercept Computer Interface (ICP) package 143.

The following packages are also required:

- Automatic Call Distribution Package A (ACDA) package 45
- Message Center (MWC) package 46
- Auxiliary Processor Link (APL) package 109
- International Supplementary Features (SUPP) package 131
- Flexible Feature Codes (FCC) package 139
- Flexible Tones and Cadences (FTC) package 125

To use the ICP Flexible DN length, DN Expansion (DNXP) package 150 is required.

To be able to use ICP in a network environment the following packages are needed: Integrated Services Digital Network (ISDN) package 145; 1.5 Mbit Primary Rate Access (PRA) package 146; and Network Attendant Service (NAS) package 159.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Allow or deny an intercept attendant to dial an extension DN from the Intercept Computer Terminal.

2 LD 15 – Set the minimum and maximum switchhook flash time required when using package 131.

3 LD 21 – Print Intercept Computer Dial from directory system information.

LD 15 – Allow or deny an intercept attendant to dial an extension DN from the Intercept Computer Terminal.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	ICP	Intercept Computer data block.
...		
- ICP	(NO) YES	Intercept Computer.
...		
- ICPD	(0)-9	ICP Padding digit.
- ICTD	(NO) YES	Intercept Computer Treatment Dial from directory. This prompt allows an intercept attendant position to dial an extension DN from the Intercept Computer Terminal. It is only prompted if ICP is set to “YES”.

LD 15 – Set the minimum and maximum switchhook flash time required when using package 131.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	TIM	Timers data block.
...		
- FLSH	xxx yyy	Switch Hook Flash timer.

LD 21 – Print Intercept Computer Dial from directory system information.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	CDB	Customer data block.
...		
ICPD	(0)-9	ICP Padding digit.
ICTD	(NO) YES	Intercept Computer Treatment Dial from directory. It is only prompted if ICP is set to “YES”.
FLSH	xxx yyy	Switch Hook Flash timer.

Feature operation

ICTD = NO

When this feature is not activated, there is no change in attendant operations.

ICTD = YES

When this feature is activated, instead of dialing from the console, it is possible for an attendant to press a single key on the Intercept Computer Keyboard.

ICTD = YES operation examples

Attendant Console is Idle

The Attendant Console is idle, all lamps are dark, the display is blank, and the Release key is lit. On the ICT the attendant types the name of the called party. The ICP database is scanned to get information about this person. The information, including extension DN 4004, is then displayed on the screen.

After the attendant presses the Dial from Directory key on the ICT keyboard, the Release lamp is dark, the Loop lamp for loop “0’ is lit, and the SRC lamp on loop “0” is slowly winking. The Attendant Console display shows DN 4004. Set 4004 is ringing.

The Attendant Console was idle when dialing was performed from the ICP computer (the call was handled as if it was initiated from the Attendant Console).

Attendant Console has Established a Call on SRC

The attendant is talking with the SRC party (DN = 4002 and ATDN is displayed), loop key “0” is lit, the SRC lamp is lit, and the DEST lamp is dark. The SRC party desires to be extended to party A. On the ICT the attendant types the name of party A. The ICP database is scanned to get information about this person. The information, including extension DN 4004, is then displayed on the screen.

After the attendant presses the Dial from Directory key on the ICT keyboard, the DEST lamp is slowly winking, the Loop lamp for loop “0” is still lit, the SRC lamp on loop “0” is still lit. The Attendant Console display shows DN 4004. Set 4004 is ringing.

The attendant was connected to the SRC party (DN 4002) when dialing from the ICP computer (the call was handled as if it was initiated from the Attendant Console).

Attendant has Call on Hold

The attendant is talking to SRC (DN 4002) and DEST (DN 4004) on loop “0”, and then puts the call on hold by pressing another Loop key, or by pressing the hold key and an idle Loop key. Loop lamp “0” is now winking; the new loop key is lit. The display is cleared.

From the ICT the attendant has typed the name of the party to be called. The ICP database is scanned to get information about this party. The information, including extension DN 4009, is then displayed on the screen.

After the attendant presses the Dial from Directory key on the ICT keyboard, the SRC lamp for this loop is winking. The Attendant Console shows DN 4009. Set 4009 is ringing.

A new loop key was selected before dialing from the ICP; the held call was not affected by this operation (the call was handled as if it was initiated from an idle loop key).

Idle Attendant Dials from Both the Attendant Console and ICT

The attendant is idle, all lamps are dark, the display is empty, and the Release key is lit. On the ICT the attendant types the name of the called party. The ICP database is scanned to get information about this person. The information, including extension DN 4009, is then displayed on the screen (information that this person could be radio paged using “*81*” is also displayed).

The attendant desires to page this person, and dials an RPAX FFC code from the Attendant Console. The Release lamp gets dark, the Loop “0” lamp gets lit, and the SRC lamp on loop “0” is slowly winking. The Attendant Console display shows RPA FFC “*81*”.

After the attendant presses the Dial from Directory key on the ICT keyboard, the DN sent from the ICP is now displayed after the RPA FFC. The paging has started and ringback tone is provided. Two dialing phases have been handled: dialing the FFC code from the console; and adding the DN from the ICT (the call was handled as if it was initiated entirely from the Attendant Console).

Attendant is Connected to DID/CO on SRC

DID/CO Releases before Dialing from ICT

The attendant is talking with the SRC party (a DID/CO trunk); Route access code, Route member, and ATDN are displayed, Loop key "0" is lit, SRC lamp is lit, and DEST lamp is dark. The SRC wants to be extended to party A. On the ICT the attendant types the name of party A. The ICP database is scanned to get information about this person. The information, including extension DN 4004, is then displayed on the screen.

The SRC goes on-hook, then the attendant presses the Dial From Directory key on the ICT keyboard. The DEST lamp is dark, Loop lamp "0" is still lit, and the SRC lamp on loop "0" is now winking. The Attendant Console display only shows DN 4004 (as SRC). Set 4004 is ringing. The DID/CO trunk is disconnected.

When the DID/CO trunk disconnects, the call dialed from the ICP will appear as a new call started from an idle Attendant Console (the call was handled as if it was initiated from an idle Attendant Console).

DID Releases after Dialing from ICT

The attendant is talking with the SRC party (a DID trunk); Route access code, Route member, and ATDN are displayed, Loop key "0" is lit, SRC lamp is lit, and DEST lamp is dark. The SRC wants to be extended to party A. On the ICT the attendant types the name of party A. The ICP database is scanned to get information about this person. The information, including extension DN 4004, is then displayed on the screen.

The Attendant presses the Dial From Directory key on the ICT keyboard. Then the SRC goes on-hook. The SRC lamp is dark, Loop lamp "0" is still lit. The Attendant Console display shows the Route access code, Route member, and ATDN on the source line, and DN 4004 on the destination line. Set 4004 is ringing. The DID trunk is disconnected.

When the DID trunk disconnects, the call dialed from the ICP will remain on the DEST side (the call was handled as if it was initiated from an idle Attendant Console).

Intercept Computer Enhancements

Contents

This section contains information on the following topics:

Feature description 673
Operating parameters 674
Feature interactions 674
Feature packaging 674
Feature implementation 674
Feature operation 676

Feature description

When an intercept transfer is activated from a customer or tenant extension, it can be configured so that only external calls are forwarded to the external intercept DN (ECDN). The internal calls are forwarded to an answering machine, or the internal intercept DN (ICDN). This applies only if the extension's flexible call forward no answer DN (FDN) is not configured as an intercept position.

The answering machine must be a multi-channel machine, connected to both the system switch and Intercept Computer. The channels are 2500-type sets, defined in a group hunt list for the answering machine. The group hunt list contains 2500-type sets with a Class of Service of Intercept Computer Answering Machine Allowed (IAMA). The Pilot DN for the Group Hunt List is defined as the ICDN, allowing calls intercepted at the Intercept Computer to terminate on the answering machine.

Operating parameters

Analog (500/2500 type) telephones can be used as Automatic Call Distribution (ACD) agent sets.

The answering machine must have a 2500-type set interface to the system.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Configure Intercept Computer Answering Machine Class of Service.

2 LD 15 – Configure internal and external call DNs for Intercept Transfer.

3 LD 93 – Configure internal and external call DNs for Attendant Console groups.

LD 10 – Configure Intercept Computer Answering Machine Class of Service.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	500/2500 telephone data block.
...		
CLS	(IAMD) IAMA	ICP Answering Machine (denied) allowed. Allow a 2500 set to be a channel in the ICP Answering Machine.

LD 15 – Configure internal and external call DNs for Intercept Transfer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	ICP	Intercept Computer update
...		
- ICDN	xxxx	Internal Call DN. DN used for intercept transfer when the FDN and multi-tenant are not on intercept position. The DN is used or intercept treatment for internal calls. Up to a four-digit DN prior to Phase 8. Up to 13 digits in Phase 8 and later.
- ECDN	xxxx	External Call DN. DN used for intercept transfer when the FDN and multi-tenant are not on intercept position. The DN is used for intercept treatment for external calls. Up to a four-digit DN prior to Phase 8. Up to 13 digits in Phase 8 and later.

LD 93 – Configure internal and external call DNs for Attendant Console groups.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	a...a	Type of data block (a...a = ACG, CPG, CPGP, RACC, RACG, RCPG, TACC, TACG, TCPG, TENS, or TGEN).
...		
- ECDN	xxxx	External Call DN. DN used for intercept transfer when the FDN and multi-tenant are not on intercept position. The DN is used for intercept treatment for external calls. Up to a four-digit DN prior to Phase 8. Up to 13 digits in Phase 8 and later. Prompted with Intercept Computer Interface (ICP) package 143.
- ICDN	xxxx	Internal Call DN. DN used for intercept transfer when the FDN and multi-tenant are not on intercept position. The DN is used or intercept treatment for internal calls. Up to a four-digit DN prior to Phase 8. Up to 13 digits in Phase 8 and later.

Feature operation

No specific operating procedures are required to use this feature.

Intercept Treatment

Contents

This section contains information on the following topics:

Feature description 677
Operating parameters 678
Feature interactions 679
Feature packaging 681
Feature implementation 681
Feature operation 682

Feature description

Calls that cannot be completed because of call restrictions or dialing irregularities can be routed to a Recorded Announcement (RAN), to the attendant, or to hear overflow, or busy tone. Separate treatments can be specified for calls from the following categories of originating party:

- Telephones
- Attendants
- attendant originated
- attendant extended
- TIE trunk, or remote attendant or telephone, and
- Controlled Class of Service Allowed (CCSA) or Direct Inward Dialing (DID) trunk.

Operating parameters

When Intercept to RAN is desired, a recording device is required. A Recorded Announcement (RAN) route and at least one trunk must be defined (see the RAN feature module).

Intercept Treatment (INTR) for these types of calls can be specified in the Customer Data Block (LD 15) for the situations listed in Table 19.

Table 19
Intercept Treatment for various types of calls.

Intercept situation	Telephone	Attendant extended calls	Calling Party TIE trunk (including attendant)	CCSA/DID trunk
Access denied (ACCD)	C(O)	C(O)	C(O)	C(A)
Call to vacant number (CTVN)	C(O)	C(O)	C(O)	C(A)
Maintenance busy number, RPE failure (MBNR)	C(O)	C(O)	C(O)	C(A)
Code or toll restricted call by Toll Denied (TLD) station or TIE trunk (CTRC)	C(O)	NA	C(O)	NA
Calls to LDNs (CLDN)	C(O)	C(O)	C(O)	NA

O = overflow tone

A = intercept to the attendant

C = choice of overflow tone, attendant, or Recorded Announcement (RAN)

NA = not applicable

Note: Items in parenthesis are the default Intercept Treatments. Where an item is preceded with “C”, a choice can be made between overflow, attendant busy, or a RAN. Four entries are required for each intercept situation.

Feature interactions

Basic/Network Alternate Route Selection (BARS/NARS)

Table 20 specifies the type of Intercept Treatments (INTR) available for BARS/NARS calls, and lists the intercept situations that are possible.

Table 20
Intercept Treatment for BARS/NARS calls

Intercept situation	Originating party			
	Station or DISA	Attendant extended calls	TIE trunk (including attendant)	CCSA/DID trunk
BARS/NARS invalid (NINV)	C(O)	C(O)	C(O)	C(A)
BARS/NARS invalid translation (NITR)	C(O)	C(O)	C(O)	C(A)
BARS/NARS restricted (NRES)	C(O)	C(O)	C(O)	C(A)
BARS/NARS blocked (NBLK)	C(O)	C(O)	C(O)	C(A)

O = overflow tone

A = intercept to the attendant

C = choice of overflow tone, attendant, or Recorded Announcement (RAN)

Note: Items in parenthesis are the default Intercept Treatments. Where an item is preceded with “C”, a choice can be made between overflow, attendant busy, or a RAN. Four entries are required for each intercept situation.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The NARS blocking treatments that can be defined through the Intercept Treatment feature are applicable to a DPNSS1 UDP network.

Flexible Feature Codes

If Intercept Treatment has been specified for a call to a vacant number (CTVN), the Digit Display (DDs) on the Attendant Console is affected by Flexible Feature Codes (FFCs). If no FFC has been defined, the dialed digits are displayed up to and including the first digit that fails to match any Directory Number (DN). If one or more FFCs have been defined, the dialed digits are displayed, up to and including the first digit that fails to match any FFC.

Ring Again on No Answer

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

Source Included when Attendant Dials

If the attendant dials a destination which is intercepted, the source remains included in the call.

Teletype Terminal Access Control in Multi-customer Environment

The Intercept Computer (ICP) feature uses maintenance LD 51 to update the system with the intercept service interface information that it stored. This overlay logs off after five minutes if no messages have been received from the Intercept Computer. This five-minute period takes precedence over the value entered in response to the LOUT prompt in LD 17. If this value is less than five minutes, the system will wait for five minutes before logging off.

Total Redirection Count

Intercept treatment is not given if a call is a Network Automatic Call Distribution (NACD) ACD call, if a call is a Central Office trunk in Night Service (specific treatment is given rather than customer-defined intercept treatment), or if the call is a data call (overflow tone is automatically given).

Trunk Barring

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

When an Originating Trunk Connection (OTC) attempts a trunk connection to a route that is restricted by its Access Restricted Table, the connection is not allowed. The intercept treatment specified in the Customer Data Block is applied.

Virtual Network Services

Intercept treatment applied to Virtual Network Service calls is configured as for TIE trunks.

Feature packaging

This feature requires Intercept Treatment (INTR) package 11.

Feature implementation

LD 15 – Change customer's Intercept Treatment for various call types.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	INT	Intercept treatment options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
INTR	(NO) YES	Allow changes to intercept treatments.
- ACCD	(OVF OVF OVF ATN)	Default Intercept Treatment for calls to access-denied numbers.
- CTVN	(OVF OVF OVF ATN)	Default Intercept Treatment for calls to vacant numbers.
- MBNR	(OVF OVF OVF ATN)	Default Intercept Treatment for calls to maintenance busy numbers.
- CTRC	(OVF NAP OVF NAP)	Default Intercept Treatment for a code or toll restricted call by a toll restricted station or TIE trunk.

- CLDN	(NAP OVF NAP NAP)	Default Intercept Treatment for calls to a Listed DN.
- NINV	(OVF OVF OVF ATN)	Default Intercept Treatment for BARS/NARS invalid calls.
- NITR	(OVF OVF OVF ATN)	Default Intercept Treatment for BARS/NARS invalid translation calls.
- NRES	(OVF OVF OVF ATN)	Default Intercept Treatment for BARS/NARS restricted calls.
- NBLK	(OVF OVF OVF ATN)	Default Intercept Treatment for BARS/NARS blocked calls.
- - RANR	0-511 0-127	RAN Route number for intercepted calls For Large Systems For Small Systems and Succession 1000 systems

Feature operation

No specific operating procedures are required to use this feature.

Intercept Treatment Enhancements

Contents

This section contains information on the following topics:

Feature description 683
Operating parameters 684
Feature interactions 684
Feature packaging 684
Feature implementation 684
Feature operation 684

Feature description

The following three intercept treatments are added for Multifrequency Compelled (MFC) Signaling:

MFC Call to Vacant Office Code

This treatment is used when a VACO level 1 signal is received from the far end.

MFC Call to Vacant Number Code

This treatment is used when a VACC level 2 signal is received from the far end.

MFC Congestion

This treatment is used when a CONG level 1 or 2 signal is received from the far end.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Dependency:

- Multifrequency Compelled Signaling (MFC) package 128.

Feature implementation

LD 15 – Modify the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	INT	Intercept Treatment options
...		
- MFVO	OVF ATN RAN NAP BSY	MFC Call to Vacant Office. Four entries are required; Default = OVF, OVF, OVF, ATN.

Feature operation

No specific operating procedures are required to use this feature.

International Meridian 1

Contents

This section contains information on the following topics:

Feature description 685
Operating parameters 686
Feature interactions 687
Feature packaging 688
Feature implementation 688
Feature operation 690

Feature description

International Meridian 1 is a feature that implements a number of significant changes to system architecture, packaging, power and performance. It consists of:

- the reformatting of terminal numbers (TNs)
- changes to call processing (to support Superloop)
- system performance improvements, and
- the support of future telephones for the system.

32 TNs (for instance, 16 integrated voice and data sets) will be supported by each Extended Line Card (XDLC) card. Each shelf can support a maximum of 16 cards.

Superloop provides an increase in traffic capacity by implementing 120 time slots for each Extended Network (XNET) Card, combining loops into groups of four, and sharing resources across the four loops. Each Extended Peripheral Equipment Controller can support between one half and four Superloops, regardless of combination, in XNET Card to DS-30X loop configurations.

The Extended Peripheral Equipment Controller packs allow monitoring of power and control functions for individual line cards. These packs also control the ringing cadences for analog (500/2500 type) telephones (set in firmware). The packs communicate with the system by way of the XNET card, which in turn communicates directly with the system using the time slot-1 address.

The Extended Analog Line Card (XALC) collects dial pulses (during dial-pulse dialing) and, upon digit recognition, sends the digit as a message to the system.

An Extended Digital Line Card (XDLC) provides voice TNs on units 0-15, and data TNs on units 16-31.

These extended packs provide enhanced maintenance and diagnostic functions. Accompanying enhancements to the system diagnostic routines allow for the handling of this system equipment.

Operating parameters

The Extended Conference and TDS (XCT) card is not supported with Supplementary features (XCT loops cannot be configured in LD 97 if the International Supplementary Features (SUPP) package 131 is equipped).

The Extended Digitone Receiver (DTR) card is supported by the system and provides the same functions as a non-system DTR card, but with a density of eight units per card (rather than four).

An Extended Network, Peripheral Control and DTR card (XNPD card) is available, providing all the functions of the XNET, XPEC and XDTR cards on one, extended card.

The system supports the configuration of the minimum/maximum flash timing on a system basis only (non-system configuration could be done on a customer basis).

System peripherals will only be available on network-enhanced machine types.

The following features are not supported on system equipment:

- Alternative Loss Plan
- Automatic Guard Detection
- Active Feature Dial Tone
- Audible Alarm
- Malicious Call Trace Enhancement
- Off-hook Tone
- Operator Call Back
- Dial Tone Detection
- Direct Inward Dialing (DID) or Direct Outward Dialing (DOD) Interface
- Enhanced Night Service
- Loop-start Supervisory Trunks
- LOGIVOX Telephones
- Malicious Call Trace Idle
- MFE
- Reverse Dial
- Ring or Hold LED Status
- R2 MFC Signaling, and
- Variable Guard Timing.

These features are still supported on non-system equipment, as they were prior to the introduction of International Meridian 1.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure the configuration record for 16-Button Dual-tone Multifrequency (DTMF) detection.

2 LD 13 – Configure DTR, TDET, and DTD cards for this feature.

3 LD 97 – Configure system parameters for peripheral equipment in configuration record 2.

LD 17 – Configure the configuration record for 16-Button Dual-tone Multifrequency (DTMF) detection.

Prompt	Response	Description
REQ	CHG	Request.
TYPE	PARM	Change system parameters.
...		
-ABCD	(NO) YES	16-tone DTMF operation enabled.

LD 13 – Configure DTR, TDET, and DTD cards for this feature.

Prompt	Response	Description
REQ	aaa	Request (aaa = CHG, END, MOV, NEW, or OUT)
TYPE	a...a	Type of data block (a...a = DTD, DTR, MFC, MFE, MFK5, MFK6, MFR, TDET, CMOD or XTD)

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
POLR	a...a	Polarity of LED messages for DTD (a...a = (NORM) or REV)
XTDT	(0)-7	Extended Tone Detector Table number.
-DTO	(NO) YES	Dial Tone Detection Only.
CDEN	a...a	Card Density (aa = SD, DD, or 4D)
TOTN	l s c u c u	To Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 97 – Configure system parameters for peripheral equipment in configuration record 2.

Prompt	**Response**	**Description**
REQ	CHG	Change.
TYPE	SYSP	System parameters.
...		
INTN	(NO) YES	µ-Law. A-law.
CODE	(0)-3	Used by Network Card firmware. 0 is the only valid entry (1-3 are reserved for future use).
CONT	1-(4)-15	Respond to the CONT prompt with the continuity error threshold value between 1 and 15 (the default is 4).
CRCF	1-(4)-15	Respond to the CRCF prompt with the CRC failure threshold value between 1 and 15 (the default is 4).
FLSH	xxx yyyy	Switch hook flash timing when International Supplementary Features (SUPP) package 131 is equipped. Minimum and maximum switch hook flash timer in milliseconds for analog (500/2500 type) telephones, where: xxx = 21-(45)-768, and yyyy = xxx value-(896)-1275.

Feature operation

No specific operating procedures are required to use this feature.

Inventory Reporting

Contents

This section contains information on the following topics:

Feature description 691
Operating parameters 699
Feature interactions 708
Feature packaging 708
Feature implementation 709
Feature operation 709

Feature description

The Inventory Reporting feature provides an automated tool for customers and support personnel to produce a hardware inventory report on the system. This report lists cards and telephones installed on the system, or configured in software.

You can use any TTY device that provides access to LD 117 to use this feature.

Optivity Telephony Manager (OTM) has a graphical user interface that supports the Inventory Reporting feature. For information related to the Inventory Reporting feature and OTM, please refer to *Optivity Telephony Manager: System Administration* (553-3001-330).

Generate Inventory files

The system can generate two separate Inventory files. The first file contains the Inventory Reporting information for all cards that are inventoried in the Card Inventory file. The second file contains the Inventory Reporting information for all telephones.

You can generate an Inventory file in LD 117. See Table 21 for a list of commands and their descriptions.

Table 21
Inventory Reporting generate commands

Command	Description
INV GENERATE ABORT	Abort all Inventory generations.
INV GENERATE ALL	Begin generating both Card and telephone Inventory files.
INV GENERATE CARDS	Begin generating Card Inventory file.
INV GENERATE SETS	Begin generating telephone Inventory file.

The two Inventory files vary slightly in their format. The first record in both types of Inventory files is the file header. The file header contains a time stamp that indicates when the Inventory process started, and when it finished. Following the time stamp, the number of records collected during Inventory appears. See Table 23 on page 694, or Table 24 on page 695, for an example of a file header.

Both Inventory files contain up to 32 bytes of Identification Programmable Read-Only Memory (ID PROM) information for each inventoried card or telephone that is physically present. The 32 bytes are actually 32 ASCII characters representing different data elements. See Table 22 for more information.

Table 22
ID PROM information

Data Element Name	Maximum Number of Characters
Product Engineering Code (PEC)	08
Color (numeric representation)	02
Release	02
Blank	01
Product Serial ID	12
Blank	01
Other (Free Field)	06

Card Inventory files

Following the file header, each record of the Card Inventory file contains:

- the card type
- the card TN, which contains:
- loop, shelf, and card numbers for IPE modules
- loop number for Network modules
- Core and slot numbers for Core cards
- 32 bytes of ID PROM information

Note: The Small System's card TN only includes the card number.

See Table 23 for an example.

Table 23
Card inventory example for Meridian 1 Option 61C

Card inventory:
17 8 1999 11 5 27, 17 8 1999 11 5 40, 15
CP , 0 14, NT9D19CA 03 NNTM1830TVFK
CCNI , 0 12, NT6D65AA 08 NNTM18304UY9
CMDU, 0 0, NT6D64AB 01 NNTM183227YT
CONF , 17, <Unavailable>
DTR , 004 0 00, NT8S16AB 03 NNTM18310C7D0000000
....

Note: <Unavailable> indicates the ID PROM information is not available because the card is not physically present.

Set Inventory files

Following the file header, each record of the Telephone Inventory file contains:

- the telephone type
- the telephone's TN (loop, shelf, card, and unit numbers)
- 32 bytes of ID PROM information
- the device's descriptor information (DES field in LDs 10 and 11)
- the primary DN

Note: The Small System or Succession 1000 system telephone's TN only includes a card number and a unit number.

See Table 24 for an example.

Table 24
Telephone inventory example for Small Systems and Succession 1000 systems

Set inventory:
17 8 1999 10 42 44, 17 8 1999 10 42 45, 4
2616, 08 01, M2616 NT2K16XC 35 01 69409A, RODNEY, 1000
2006, 08 01, M2006 NT2K05XH 93 10 C10C19, CHRIS , 1100
2008, 08 02, M2008 NT9K08AD 03 03 945272, DEBBIE, 1200
2616, 08 03, M2616 NT2K16XD 35 01 CC9C98, DANNY , 1300
2616, 02 10, <Unavailable>, TROY , 5902
....

Note:
<Unavailable> indicates the ID PROM information is not available because the telephone is not physically present, or is disabled (“DSBL” in LD 32).

Backup files

The system keeps a current file and a backup file for each Inventory file. Each request to generate an Inventory file causes the previous current file of the same type to become a backup file. The system can use the backup file in the event that the generation of a new file is not successful.

Files in use

If you request to generate an Inventory file while the system is generating that file, you will receive a "Card (or Set) file is Generating, try again later" message.

Abort generation

You can abort the generation of Inventory files. If there is any generation of a Card or Set Inventory file when you execute the **INV GENERATE ABORT** command, the system stops gathering data for the Inventory generation.

If the system receives an abort request and there is no activity on a file, the request is rejected, and you will receive a "No generation to abort" message.

Midnight Routine

To schedule Inventory Reporting for the virtual midnight routine, use the commands in LD 117. See Table 25 for a list of commands and their descriptions.

Table 25
Inventory Reporting midnight routine commands

Command	Description
INV MIDNIGHT ALL	Schedule Card and Telephone Inventory file generation.
INV MIDNIGHT CARDS	Schedule Card Inventory file generation.
INV MIDNIGHT OFF	Unschedule Card and Telephone Inventory file generation.
INV MIDNIGHT SETS	Schedule Telephone Inventory file generation.
INV MIDNIGHT STATUS	Print state of virtual midnight routine schedule of Inventory Reporting.

Printing Inventory files

The process of generating an Inventory file is separate from the process of printing an Inventory file on the TTY. You can print an Inventory file on the TTY from the CLI in LD 117. See Table 26 on page 697 for a list of commands and their descriptions.

Table 26
Inventory Reporting print commands

Command	Description
INV PRT	Print out the status of the Inventory feature
INV PRT ALL	Print out both the card and the Telephone Inventory files
INV PRT CARDS	Print out the Card Inventory file
INV PRT SETS	Print out the Telephone Inventory file
INV PRT STATUS	Print out the status of the Inventory feature

When you execute the print command, the selected Inventory file is scrolled onto the TTY. When you print an Inventory file, the system automatically selects the current file (rather than the backup file). Printing an Inventory file cannot be scheduled by the system.

Once the printing process has started, you can abort it by exiting out of LD 117 using four asterisks (****).

There is no notification of completion for printing out an Inventory file onto the TTY.

Inventory Reporting status

There are two commands that can be used to query the Inventory Reporting feature:

- INV PRT
- INV PRT STATUS

The response to a status query contains two responses, one for the Card Inventory file and another for the Set Inventory file. You only need to make a single request for both files.

The response indicates whether each file is:

- OK (Idle)
- DOWNLOADING
- BUSY
- GENERATING

Note: Only the status of the current file(s) is provided. The status of the backup file cannot be obtained using the status command.

See Table 27 for a list of status responses and their descriptions.

Table 27
Inventory Reporting status responses

Response	Description
BUSY	When the Inventory file is in use.
DOWNLOADING	When the Inventory file is being downloaded.
GENERATING	When the system is generating the Inventory file.
OK	When there is no activity using the Inventory file(s).

Operating parameters

When a telephone is installed, but not configured in software, the system has no record of the telephone, and therefore, will not be inventoried. A telephone that is installed, but configured in software as a different type of telephone, may not be included in the inventory file.

The Inventory Reporting feature can only report ID PROM information from cards and telephones that are physically present. If a card or telephone is configured in software, but is not present in the system, then the ID PROM information will not be inventoried.

Any new cards, or existing cards, that emulate another type of card in the system, when inventoried is noted to have the card type of that emulated card, and not its correct card type. The correct engineering code and vintage of the actual card is listed in the Card ID PROM information, if available.

When there is a dual processor (redundant) system, the Inventory Reporting feature will not incorporate the standby processor and associated cards (Central Processor and Core Network Interface cards) in the card report.

Inventoried cards

Table 28
Card types are included in the Card Inventory file

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
BRSC	Basic Rate Signaling Concentrator	NT6D72	AA	North America
CMDU	Core MultiDrive Unit	NT6D64	AA	North America
CCNI	Core to Network Interface	NT6D65	AA	North America
COT	CO Trunk	NT5K93	AA, AB, BA, BB	Global
CP	CP68030/24MB, Call Processor	NT6D66	AA	Global
CP	CP68030/48MB, Call Processor	NT9D66	DA	Global

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
CP-2	CP68040/48MB, Call Processor	NT9D19	AA, AB	Global
CP-2	CP68040/64M/32M, Call Processor	NT9D19	HA	Global
CP-2	CP68040/64MB, Call Processor	NT9D19	CB	Global
CP-2	CP68040/96MB, Call Processor	NT9D19	HB	Global
CP-3	CP68060/112MB, Call Processor	NT9D10	JA	Global
CP-3	CP68060/48MB, Call Processor	NT9D10	AA	Global
CP-3	CP68060/64MB, Call Processor	NT9D10	CA	Global
CP-3	CP68060/80MB, Call Processor	NT9D10	EA	Global
CP-3	CP68060/96MB, Call Processor	NT9D10	HA	Global
CP-4	CP4 Call processor	NT5D03	AA-UA	Global
CPP	System Utility Card	NT4N67	AA	Global
CPP	System Utility Transition Card	NT4N68	AA	Global
CPP	LED/LCD Display Panel	NT4N71	AA	Global
CPP	CCNI Card	NT4N65	AA	Global
CPP	CPU Card	A0810496	N/A	Global
CPU	68K Processor Card - Card Option CPU	NTAK14	AA, BA	North America
CT2	Line Card, Mobility	NTCK93	AA	International
DDP	Digital Trunk, DTI/PRI, Double	NT5D12	AF	North America
DDP2	Digital Trunk, DTI/PRI, Double E1	NT5D97	AB	International
DID	DID Trunk	NT5K84	AA, AB, BA	International

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
DID	DID Trunk, on board PPM, extended three wire	NT5K60	AA, AB	International
DID	DID Trunk, on board PPM, on board detection	NT5K36	AB, BA	International
DID	Trunk Card	NT5D28	AA	India
DPRI	Digital Trunk, PRI2, Double E-1	NTCK43	AC	International
DTI/PRI	1.5 MB DTI/PRI	NTAK09	DA	North America
DTI2	CIS Trunk for Small Systems and Succession 1000 systems	NTCG02	BA, BB	CIS
DTI2	CIS Trunk for Large Systems	NTCG01	BA, BB	CIS
DTI2	2.0 MB DTI	NTAK10	DC	International
DXUT	Universal Trunk	NT5D31	AA	International
DXUT	Universal Trunk, Extended	NTAD14	EA, DA	International
EIMC	Embedded Intelligent Mobility Controller	NT7R01	CA	North America
EXALCC	Analog Line Card	NTRA08	AA, AB, BA	China
EXUTAP-1	Universal Trunk, Busy Tone detect Trunk, 400Hz	NTRA26	AA	Global
EXUTAP-2	Universal Trunk, Busy tone detect Trunk, 425Hz	NTRA26	BA	Global
EXUTC	Universal Trunk, Extended	NTRA10	AA, AB	China
EXUTJ	Universal Trunk	NT8D14	DA	Japan
EXUTJ	Universal Trunk, Extended	NT5D15	AA	Japan
FXNET	Fiber Extended Network	NTIP61	BA	Global
FXPEC	Fiber Extended Peripheral Equipment	NTIP62	CA	Global

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
IODU	I/O Disk Unit	NT5D20	BA	Global
IODUC	I/O Disk Unit w/ CD-ROM	NT5D61	AA, AB, BA	Global
IOP	I/O Processor	NT6D63	BA	Global
ITG	24 Ports ISDN	NTZC44	AA, BA	Global
LCI	Local Carrier Interface	NT7R51	AC, AD	North America
LE1	Line Side E1	NT5D33	AA, AB	International
LT1	Line Side T1	NT5D11	AB, AC	North America
MGATE	Meridian Mail Gateway - IPE version of MCE	NTRH14	AA	North America
MGATE	Meridian Mail Gateway - MM	NTRB18	AA	North America
MGATE	Meridian Mail Gateway - Tower version of MCE & MM	NTRB18	AA	North America
MICA	Integrated Call Assistant	NT5G11	AA	Global
MICB	Integrated Conference Bridge Base	NT5D51	AA, AB, AC	North America
MIRAN	Meridian Recorded Announcement	NTAG88	AA	North America
MISP	Multi-Purpose ISDN Signaling Processor	NT6D73	AA	North America
MXC	MicroSystem Transcoder	NTEX80	AA	North America
NCE	Fiber in Junctor Interface Motherboard	NTRB3301	N/A	Global
NCE	Fiber in Junctor Interface Jumper Daughterboard	NTRB3303	N/A	Global
NCE	3 Ports CCNI	NTRB34	AA	Global
PRI2	2.0 MB PRI	NTAK79	BC	International

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
PRI2	2.0 MB PRI	NTBK50	AA	International
RCI	Remote Carrier Interface	NT7R52	AC, AD	North America
SILC	S/T Interface Line Card	NT6D70	AA, BA	North America
TMDI	T1 Multi-purpose digital interface for Small Systems	NTRB21	AA	North America
UILC	U Interface Line Card	NT6D71	AA, AB	Global
VPS	Voice Processing Application Server	NTAG36	AA	North America
XALC	Analog Line Card	NT8D03	AA-Ak	North America
XALCC	Analog Line Card	NTRA05	AA	Global
XCOT	CO Trunk	NT5K82	AA, AB	Global
XCOT	CO Trunk	NT5K90	AA, AB, BA, BB,	Global
XCOT	CO Trunk	NT5K99	AA, BA	Global
XCOT	Trunk Card	NT5D29	AA	India
XCOTI	CO Trunk	NTRA29	AA	Global
XDAC	X-Calibur Data Access	NT7D16	AA	North America
XDID	DID Trunk, Extended	NT5K36	AA	International
XDID	DID Trunk, Extended	NT5K84	HA	International
XDID	DID Trunk, Extended	NTAG04	AA	International
XDID	DID Trunk, Extended	NTRA28	AA	International
XDID	DID Trunk, Extended Flexible	NT5K17	AB, BA, BB	International
XDID	DID/LDR Trunk, Extended	NTCK22	AA, AB	International
XDLC	Digital Line Card	NT8D02	GA	Global

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
XDTMF	Extended DTMF Receiver	NTRA11	AA	International
XDTR	Extended DTMF Receiver	NT8D16	AB	North America
XDTRC	Extended DTMF Receiver	NTRA11	AA	China
XEM	E & M Trunk Leads PCBA, Extended	NT8D15	AF, AH, AA	North America
XEMC	E & M Trunk, Extended	NTRA03	AA	China
XFALC	Analog Line Card, Flexible High Voltage	NT5K96	EA, HA, JA, JB, KA, NB	Global
XFALC	Analog Line Card, Flexible High Voltage	NT5K02	AA, AB, AC, DA, DB, EA, EB, JA, JB, JC, KA, KB, LB, LC, LD, MA, MB, MC, NB, NC, PA, PB, PC, QA, QB, QC, SA, SB, TA, TB	Global
XFALC	Analog Line Card, Flexible High Voltage, Message‘	NT5K96	MA, MB, NB, PB, SA, TA	Global
XFALCC	Analog Line Card, Message Waiting	NTRA04	AA	Global
XFCOT	CO Trunk, Extended Flexible PPM	NT5K18	AA, AB, BA, BB	Global
XFCOT	CO Trunk, Extended Flexible PPM	NT5K61	AA	Global
XFCOT	CO Trunk, Extended Flexible PPM	NT5K82	BA, BB, CA, HA	Global
XFCOT	CO Trunk, Extended Flexible PPM	NTAG03	AA, AB	Global

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
XFCOT	CO Trunk, Extended Flexible PPM, 4 unit	NT5K71	AA, AB	Global
XFCOT	CO Trunk, Extended Flexible PPM, 8 unit	NT5K70	AA, AB	Global
XFCOT	CO Trunk, Extended Flexible	NTCK16	AA, BA, BC, BD, BE	Global
XFCOT	CO Trunk, Extended Flexible PPM	NTCK18	AA	Global
XFCOT	CO Trunk, Extended Flexible PPM	NTCK24	AA	Global
XFCOT	CO/FX/WATs Trunk	NT9C14	AA, BA	Global
XFEM	E & M Tie Trunk, Wire with recorded ann, & Paging	NT5K83	BA, BB, CA, CB	International
XFEM	E & M Tie Trunk, 4 Wire with recorded ann. & Paging	NT5K19	BA, BB,	International
XFEM	E & M Tie Trunk, Extended Flexible	NT5K50	AA	International
XFEM	E & M Tie Trunk, Extended Flexible	NT5K83	AA, AB, DA, DB, FA, GA, HA, KA, LA	International
XFEM	E & M Tie Trunk, Extended Flexible, 4 unit	NT5K72	AA	International
XFEM	E & M Trunk, Extended Flexible	NT5K19	AA, AB, AC	International
XMFC	Extended Multi-Frequency Compelled Sender Receiver	NT5K21	AA	International
XMFR	Extended MF Receiver	NTAG26	AA	International
XMLC	Message Waiting Line Card	NT5D49	AA	International
XMLC	Message Waiting Line Card	NT5D09	AA, BA, LA, PA	International

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
XMWLC	Analog Line Card, Message Waiting	NT8D09	BA, AL	North America
XNET	Extended Network	NT8D04	BA	Global
XOPS	Analog Line Card	NT1R20	AA	North America
XOPSC	Analog Line Card	NTRA06	AA, AB	Global
XPEC	Ext Peripheral Equipment Controller 2 Superloop	NT8D01	BD	Global
XPEC	Ext Peripheral Equipment Controller 2/4 MB	NT8D01	DA	Global
XPEC	Ext Peripheral Equipment Controller 2MB	NT8D01	EA	Global
XPEC	Ext Peripheral Equipment Controller 4 MB	NT8D01	CA	Global
XPEC	Ext Peripheral Equipment Controller 4 Superloop	NT8D01	BC	Global
XSM	Extended System Monitor	NT8D22	AC	Global
XTD	Extended Tone Detector	NT5K48	AA-HA	International
XUT	Universal Trunk	NT8D14	BB, BC	North America
XUTC	Universal Trunk	NTRA02	AA	China

The following card types are not included in the Card Inventory file:

- TTY or PC cards
- Power Supply
- Any non-Nortel Networks (third-party) cards including those designed to simulate included cards.

Inventoried telephones

The Telephone Inventory file includes the following telephones:

- M2006
- M2008
- M2016
- M2216
- M2616
- M3110
- M3310
- M3820
- M3901
- M3902
- M3904
- M3905

The Telephone Inventory file does not include ID PROM information for the following telephones:

- Sl-1 telephones
- 500/2500 telephones
- Other digital telephones or any non-Nortel Networks (third-party) telephones, including those designed to simulate included telephones.

Data units

A data unit's TYPE is listed in the Telephone Inventory file as the TYPE of the telephone that it is attached to, not as a data unit. Data units can be identified in the Telephone Inventory file by determining which TN is assigned to the data unit, or by its descriptor information (DES field in LDs 10 and 11).

The Telephone Inventory file does not include ID PROM information for the following data units:

- Data units on:
  - M2006
  - M2008
  - M2016
  - M2616
  - M2216
  - M390X
  - M3110
  - M3310
  - M3820
- Sl-1 data units.
- 500/2500 data units.
- Other digital data units.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Italian Central Office Special Services

Contents

This section contains information on the following topics:

Feature description 711
Operating parameters 711
Feature interactions 712
Feature packaging 713
Feature implementation 713
Feature operation 714

Feature description

This feature allows callers to access “1xx” special services of the Italian Central Office (CO). The special services are accessed by dialing a Flexible Feature Code (FFC) of up to four digits in length. This FFC is configured in LD 57.

This feature is available on Meridian 1 proprietary telephones, analog (500/2500 type) telephones, as well as Attendant Consoles.

Operating parameters

This feature can only be activated by callers on the same node as the Central Office trunk; it is not supported on Integrated Services Digital Network (ISDN), Digital Private Network Signaling System 1 (DPNSS1), or other trunks.

This feature is only allowed for a simple call, and cannot be accessed in consultation state.

As a result, if the attendant makes a “1xx” service call on the source side, a call cannot be made on the destination side; therefore, the special service call cannot be extended or transferred.

Outgoing digits are outpulsed according to the trunk Class of Service, dial pulse (DIP) or digitone (DTN).

An attendant or a set accessing a special “1xx” service cannot establish a conference by pressing the Conference key or Loop key.

Analog trunks on Small Systems and Succession 1000 systems are not supported.

Feature interactions

The following features are not allowed if a special “1xx” service is being accessed:

- Multi-Party Operations
- Conference
- Transfer
- Call Join, and
- Consultation Hold (on 500 and 2500 sets).

The following features are not allowed from an attendant to a set making a special “1xx” service call:

- Priority Override
- Attendant Break-in
- Attendant Barge-in, and
- Busy Verify.

Call Detail Recording

The start timing on the Call Detail Recording record corresponds to the seizure of the Central Office trunk or to the answer signal, when received.

Periodic Pulse Metering

Periodic Pulse Metering pulses are received from the Central Office according to the charge of the accessed service, and are collected and stored as per normal procedures.

Switchhook flash

A switchhook flash is ignored while a special "1xx" service is being accessed.

16-Button Digitone/Multifrequency Operation

The special service FFC is not supported on the ABCD keys of 16-button DTMF sets.

Feature packaging

The following packages are required:

- End-to-end Signaling (EES) package 10
- 2 Mbit Digital Trunk Interface (DTI2) package 129 to support digital trunks
- International Supplementary Features (SUPP) package 131
- Trunk Hook Flash 157; and Flexible Feature Codes (FFC) package 139

Feature implementation

LD 57 – Configure the Flexible Feature Code required to access "1xx" special services.

Prompt	Response	Description
REQ	CHG	Change
TYPE	FFC	Flexible Feature Code data block
...		
CODE	a...a	FFC to be changed
...		

ITXX	1-4	FFC to access "1xx" special services.
RTXX		The CO route number for the “1xx” special service, prompted only if ITXX has been configured.
	0-512	For Large Systems

Feature operation

Dial the Flexible Feature Code (up to four digits in length) that was configured in LD 57 to access “1xx” special services.

Only the **Rls** and **Hold** keys may be activated during a call to a special “1xx” service.

Italian Periodic Pulse Metering

Contents

This section contains information on the following topics:

Feature description 715
Operating parameters 716
Feature interactions 716
Feature packaging 717
Feature implementation 717
Feature operation 717

Feature description

A new vintage 2 Mbps Digital Trunk Interface (DTI2) card is introduced with this feature. The Italian Periodic Pulse Metering (PPM) feature enables this new DTI2 card to count PPM pulses on Italian DTI2 trunks.

In Italy, a pulse on the A bit while the B bit is zero (P0UU) is considered a valid PPM pulse. However, a pulse on the A bit while the B bit is one (P1UU) should not be considered a valid PPM pulse.

When the DTI2 card detects that a pulse on the PPM bit (A in Italy) has met all PPM timing requirements, the DTI2 card checks to see if the Italian PPM feature is enabled. If so, the state of the B bit is also checked. At this point, the PPM count will be incremented (in the card) only if the B bit is zero. Using the Italian PPM option, the new card no longer reports the P1UU case as a PPM pulse. With this feature enabled all state changes with B bit set to one (for example, P1UU) are reported immediately by the DTI2 card. This allows the main Central Processing Unit (CPU) to recognize Italian Release Control pulses.

Operating parameters

This feature is not supported on Small Systems and Succession 1000 systems, since there is no XDTI2 card supporting the hardware requirements).

The feature does not work on the following DTI2 cards: QPC915A, QPC915B, QPC536A, QPC536B, QPC536C, and QPC 536D. All of these DTI2 cards, do not have the required firmware modifications.

The firmware checks whether the B bit is zero. This is hard coded in the new DTI2 cards. Other combinations are not possible (for example, it is not possible to report PPM pulses on the A bit only when the C bit is zero, and it is not possible to report PPM pulses on the A bit only when the B bit is 1).

The Italian PPM option is stored for each loop. Hence, the Italian PPM option is set the same for all channels on the loop.

Feature interactions

Call Detail Recording

This feature now allows Call Detail Recording on Italian DTI2 trunks to show the cost of the call calculated from the PPM pulses.

Periodic Pulse Metering

This feature now allows PPM pulses to be counted on Italian DTI2 trunks. The Italian DTI2 option default is set to NA (that is, not active when software prior to the introduction of this feature is upgraded). The existing operation thus continues unaffected by the new feature.

Feature packaging

This feature is packaged under the existing 2 Mbps Digital Trunk Interface (DTI2) package 129. Periodic Pulse Metering/Message Registration (MR) package 101 is required for its operation.

Feature implementation

LD 73 – Configure the Italian PPM option.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DTI2	2 Mbit digital trunk.
FEAT	LPTI	Loop timers.
LOOP	nnn	Loop number.
...		
ITPP	(NA) YES NO	Italian PPM option. If this is set, PPM pulses are only counted when the B bit is zero. NA = The DTI2 card is not capable of Italian PPM (the default). YES = Turn on Italian PPM in DTI2 card. NO = Turn off Italian PPM in DTI2 card.

Feature operation

No specific operating procedures are required to use this feature.

KD3 Direct Inward Dialing/Direct Outward Dialing for Spain

The KD3 Direct Inward Dialing (DID)/Direct Outward Dialing (DOD) for Spain feature is introduced to enable the system to meet the specifications of the Spanish signaling protocol. Prior to the introduction of the KD3 interface, the only Central Office trunk support available in Spain from a system perspective was an analog Central Office Trunk (COT) type of interface (that is, non Digital Trunk Interface (DTI) or DID/DOD).

Only KD3 to Meridian Customer Defined Network (MCDN) tandeming will be supported (no other networking protocols will be supported at this time).

The KD3 interface utilizes the following:

Digital Interface

A 2.048 Mbit digital link physical interface conforming to International Telegraph and Telephone Consultative Committee (CCITT) G700 series specifications, and whose frame and multiframe structure conform to CCITT recommendations G732 and G734, is specified.

Multifrequency Interregister Signaling

A Multifrequency Interregister Signaling protocol is used for passing certain information such as addressing and Call Class. It is similar to Multifrequency Extended (MFE), but must support both 2/5 or 2/6 frequency encoding on a system basis. It also uses different signals, and adds several new timing parameters. The new signals are mainly used to provide Class of Call information, broken down as Regular Subscriber, Special Services, National and International calls.

For more information on KD3 Signaling, please see the KD3 Signaling document contained in the IPE supplement for Spain.

Last Number Redial

Contents

This section contains information on the following topics:

Feature description 721
Operating parameters 722
Feature interactions 722
Feature packaging 726
Feature implementation 726
Feature operation 728

Feature description

Last Number Redial (LNR), which is defined on a customer and a telephone basis, allows the last number dialed by a user to be automatically stored. The stored number can be redialed by pressing a key on Meridian 1 proprietary telephones, or by dialing SPRE + 89 on analog (500/2500 type) telephones. The number is stored whether the call rings, is busy or answered, or a valid access code is dialed with the number. Only one number, composed of up to 32 digits (including access codes), can be stored at any one time. The new number overwrites the previously stored number.

If the telephone has a Digit Display (DDS), the called number is displayed.

Operating parameters

When making a call using Last Number Redial (LNR), no digits can be dialed before the stored number except Authorization, Charge Account, or Forced Charge Account codes. However, additional digits can follow the outpulsed LNR number.

The M3000 and the M2317 telephones have LNR as a local telephone (firmware) feature instead of as a system feature.

Feature interactions

AC15 Recall: Transfer from Meridian 1

Autodial and Last Number Redial are supported with the AC15 Recall: Transfer from system on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone detection is performed by the system.

Additional transfers are supported if the stored digits are outpulsed without any treatment. For example, a route is seized and the route access code is outpulsed to the far end and interpreted as a Directory Number. No dial tone detector or timer is started, so the digits are outpulsed immediately without checking the state at the far end.

Authorization Code Security Enhancement
Charge Account
Charge Account, Forced

These codes are not stored in Last Number Redial (LNR). To use these features when calling the number stored in LNR, the code must first be dialed manually. When dial tone is returned, LNR can be used to complete the dialing.

Autodial

A number dialed using Autodial will become the LNR number on all telephones, except the M2317 telephone and M3000.

Autodial Tandem Transfer

Normally, when the ADL key is pressed during the dialing stage, the ADL number will replace the Last Number Redial number. In the ATX feature, however, when the ADL key is used during the established stage, the ADL digits will not substitute the Last Number Redial number.

Automatic Redial

An Automatic Redial (ARDL) call can be activated on a number dialed using the Last Number Redial (LNK) key or by pressing the DN key twice. The ARDL number is saved as the last number redialed.

Call Forward/Hunt Override Via Flexible Feature Code

The Call Forward/Hunt Override via Flexible Feature Code and the dialed DN are stored under Last Number Redial.

Call Modification

When a Call Modification takes place at the called Directory Number, the originally dialed number and not the number reached through Call Modification is stored as the LNR. This applies to the following features:

- All Call Forward features
- Call Pickup
- Conference
- Hunting
- Integrated Messaging System (IMS) when using Operator Revert, and
- Transfer.

The stored LNR number will not be affected when making calls using the following features:

- Numbers dialed on Call Transfer or Conference
- Attendant Recall from Meridian 1 proprietary telephones (using key)
- Call Park
- Dial Intercom

- Group Call, and
- Special Services Access Codes.

Calling Party Privacy

The Last Number Redial (LNR) feature will set a Calling Party Privacy (CPP) flag in the LNR data space if the CPP was included in the last number dialed by the user. Any subsequent outgoing redialed call will send the Privacy Indicator to the far end.

Enhanced Flexible Feature Codes - Outgoing Call Barring

Barred DNs will be stored by Last Number Redial (LNR). DNs redialed using LNR are checked against the active OCB level.

OCB Flexible Feature Codes are not stored as the last number dialed.

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Last Number Redial.

Conference

When a M2317 telephone conferences in another call, goes on-hook and activates the Last Number Redial (LNR), the LNR feature redials the last number dialed during conference. However, on sets other than the M2317, LNR dials the DN dialed prior to conference.

Group Hunt

A Pilot DN will be stored as a Last Number Redial (LNR) number when it is dialed directly.

Multiple Appearance Directory Number

A last number dialed on a Directory Number (DN) with multiple appearances is stored only against the telephone from which the number was originally dialed.

Multi-Party Operations

For analog (500/2500 type) telephones, the Last Number Redial/Stored Number Redial feature can be used when normal or special dial tone is received. The last number redialed that can be stored is the first call of a consultation connection, and can be stored only after the connection is completely released.

Network Intercom

A Hot Line key cannot be redialed using the Last Number Redial feature.

Off-Hook Alarm Security

Off-Hook Alarm Security treatment may apply to these features if the ASTM expires.

Speed Call

A number dialed using Speed Call will become the LNR number on all telephones, except the M2317 and M3000.

Speed Call, System

A number dialed using a System Speed Call key becomes the Last Number Redial number on all telephones, except the M2317 and M3000. A number dialed using SPRE-activated System Speed Call becomes the Last Number Redial number on all telephones. The original Class of Service and NCOS restrictions of the telephone apply when using Last Number Redial.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Last Number Redial on an E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Transfer

When a M2317 telephone transfers a call, goes on-hook and activates Last Number Redial (LNR), the LNR feature redials the last number dialed during the transfer. However, on sets other than the M2317, LNR dials the DN dialed prior to transfer.

Feature packaging

Last Number Redial (LNR) package 90 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable or disable LNR for a customer.

2 LD 10 – Add or change LNR for analog (500/2500 type) telephones.

3 LD 11 – Add or change LNR for Meridian 1 proprietary telephones.

LD 15 – Enable or disable LNR for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(LRD) LRA	LNR (denied) allowed.

LD 10 – Add or change LNR for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(LND) LNA	LNR (denied) allowed.
LNRS	4-(16)-31	LNR size.

LD 11 – Add or change LNR for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(LND) LNA	LNR (denied) allowed.
LNRS	4-(16)-31	LNR size.
KEY	xx LNK	LNR key, where, xx = key number.

Feature operation

To automatically redial the last number dialed:

- Lift the handset or select a free Directory Number (DN).
- Press the **Last No.** or the **DN** key again.

To automatically redial the last number dialed (analog (500/2500 type) telephones):

- Lift the handset.
- Dial SPRE+89.

Limited Access to Overlays

Contents

This section contains information on the following topics:

Feature description 729
Operating parameters 732
Feature interactions 732
Feature packaging 733
Feature implementation 733
Feature operation 737

Feature description

Limited Access to Overlays allows the administrator to limit access to a configured database. It allows an administrator to define up to 100 login passwords in the configuration record (LD 17), each with its own set of access restrictions. For each Limited Access Password (LAPW), define the level of access the password provides:

- Only the Overlay numbers defined for each password can be accessed.
- Only the customer data specified can be modified by users of each password.
- Only the tenant numbers allowed can be accessed.
- Access to Print Routine LD 20 may or may not include access to the Speed Call lists.

- Access to the Configuration Record (CFN) LD 17 can be restricted to:
  - no access at all to LD 17
  - changing a user's own password only
  - full access to modify the system configuration
- With the Print Only option defined, certain users are limited to the following:
  - Access only to administration Overlays that contain print commands, and can only use the print commands in those Overlays.
  - Full access to all print routines: LDs 20-22 and LDs 81-83.
  - System commands in Traffic Overlay 02 are accessible only to users with access to all customers. Customer-defined commands are accessible according to the customer numbers defined for each password.

Only the highest level password users – Level 2 or PWD2 – can configure or change access for other passwords. These users are the administrators.

Implementing and using the LAPW feature does not interfere with the use of any existing passwords in the system. For a complete listing of the passwords currently used, refer to LD 17, prompts PWD2, NPW1, NPW2, and LD 15, prompts ATAC and SPWD in *Software Input/Output: Administration* (553-3001-311).

Each password can access up to 32 customer-tenant combinations. Each combination is defined by a number designator that includes the customer number (0-99) and the tenant number (0-511).

Each new Limited Access Password (LAPW) must be:

- any combination of numbers and letters (uppercase letters only)
- four to sixteen characters in length with no spaces
- leftwise unique, and
- different from existing passwords.

For example, acceptable passwords can include:

- JSMITH
- 0001
- 2GUEST, and
- TECHNICIAN.

System administrators using PWD1 and PWD2 in LD 17 define access to Overlays with this feature. They may also define certain command use levels within a given Overlay. For instance, the administrator can specify print only access in the Configuration record (LD 17). Any other requests generate the following system message:

SCH8836 PASSWORD HAS PRINT ONLY CLASS OF SERVICE.

After calling up an Overlay, certain commands can be restricted from use by the same password, if that password is properly defined. Trying to use those commands without the correct password is not successful – access is denied.

Logon attempts are monitored for security. Failed attempts with invalid passwords are counted and the tally is compared with a predefined threshold. If the threshold is met or passed, the entry point (TTY or terminal) is locked out for a predetermined time set in Service Change (and password protected). Access from that point is ignored by the system for the lock-out timer defined. Lock-out conditions are reported to all maintenance terminals when they occur, with a special report to the next system administrator who logs on.

The system can keep an Audit Trail to record login information. The four columns in the Audit Trail printout contain:

column 1 – DAT (date, appears at beginning of each day), or
LOG (a login record)

column 2 – aa/bb (month/day), or
cc:dd (hours: minutes)

column 3 – #ee (number associated with password)

column 4 – ff ff . . . (LD numbers accessed)

Figure 17
Example of Audit Trail printout (LD 22)

```
DAT   01/02
LOG   08:01  #03    10  11
LOG   09:32  #04    15  10  21  57  22  11  15  21
                    14  15
LOG   11:21  #99    12
LOG   16:35  PWD2   15  17
```

Only system administrators, logged in using PWD1 or PWD2, can access the Audit Trail from LD 22.

Administrators can change the size of the Audit Trail buffer, which can be from 50 to 1000 words (divisible by 50). When the buffer is full, new records overwrite the oldest information in the buffer (message OVL401 is sent to the active TTY and all maintenance TTYs). Printing the Audit Trail in LD 22 clears the buffer.

Operating parameters

The LAPW feature should only be enabled on a system with a completed Configuration record in LD 17 (a system that is already up and running). All passwords defined within the feature must be unique. Users and administrators cannot have more than one password defined for any one access configuration.

Feature interactions

Set-Based Administration Enhancements

The Set-Based Administration access passwords which are added to LAPW are subject to the same conditions as the overlay access passwords with the following exceptions:

- Set-Based Administration passwords must be numeric.

- There is no maximum number of login attempts for Administrator or Installer sets. Lockout procedures are not used.
- TTY users are not permitted to login using a Set-Based Administration password.
- Administration sets and User sets are not permitted to login using overlay access passwords.

The total number of LAPW passwords allowed, including overlay access and Set-Based Administration access, is 100.

The permissions and restrictions associated with a Set-Based Administration password used to login to an Administration set or Installer set remain unchanged throughout the login session. Thus, if a TTY user changes a Set-Based Administration password (in LD 17) while an Administration or Installer set is logged in with the same password, the permissions and restrictions associated with the session are not affected. The changes come into effect the next time a user logs in.

Feature packaging

This feature requires Limited Access to Overlays (LAPW) package 164.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Define LAPW options and passwords.

2 LD 17 – Change user's LAPW password (user must log in using current LAPW).

3 LD 22 – Check options available for LAPW passwords (administrator).

4 LD 22 – Print options for LAPW password (user).

5 LD 22 – Print contents of Audit Trail buffer (allowed if using PWD1 or PWD2).

Implementing the LAPW feature requires that you change the Configuration record (CFN) in LD 17.

LD 17 – Define LAPW options and passwords.

Prompt	Response	Description
REQ	CHG END	Change data, or terminate overlay.
TYPE	CFN PWD	Configuration Record. Gate opener.
- PWD2	xxxx	Current level 2 master password.
- NPW1	xxxx	New level 1 login password.
- NPW2	xxxx	New level 2 master password.
- LAPW	0-99	LAPW password number.
- - PWnn	dd...d <CR>	New password for “nn” above. No more changes to LAPW.
- - OVLA	xx xx xx . . .xx, ALL (XALL)	Add these overlays to the list access by password PWnn. Xnn removes the overlay.
- - CUST	0-99, ALL (XALL)	Customer number, all customers (no customers).
- - TEN	xxx xxx . . . xxx, ALL (XALL)	Tenant list for the above customer for password access. XALL removes tenant access for this password.
- - HOST	(NO) YES	Host mode.
- - OPT	aaaa (CFPA) CFPD (LLCD), LLCA (PROD) PROA (PSCA) PSCD	Password Options allowed. Changes to all LD 17 prompts (allowed) denied. Line Load Control commands (denied) allowed. Print Only Class of Service (denied) allowed. Printing Speed Call lists (allowed) denied.
- LAPW	<cr>	Stop defining passwords.
- FLTH	0-(3)-7	Failed logon attempt threshold.
- LOCK	0-(60)-270	Lock-out time in minutes.

- AUDT	(NO), YES	Audit Trail (denied) allowed.
- SIZE	(0)-65534	Word size stored in the Audit Trail buffer.
- INIT	(NO) YES	Reset ports locked out during manual INIT.

LD 17 – Change user's LAPW password (user must log in using current LAPW).

Prompt	**Response**	**Description**
REQ	CHG	Change password options.
TYPE	CFN PWD	Configuration Record. Gate opener.
- PWD2	<CR>	Level 2 master password.
- LPWD	aaaa	Logon Password for LAPW user.
- - NLPW	xx . . . x	New logon password for LAPW user.

LD 22 – Check options available for LAPW passwords (administrator).

Prompt	Response	Description
REQ	PWD	Lookup password options.
PWD2	xxxx	Level 2 master password.

Note: LAPW password options are output to the active TTY only. Options format is shown below:

FLTH	x	Failed logon attempt Threshold.
LOCK	xx	Lock-out time in minutes.
AUDT	aaa	Audit Trail allowed (denied).
SIZE	xxxx	Word size stored in the Audit Trail buffer.
INIT	aaa	Reset ports locked out during manual INIT.
PWD1	xxxx	Level 1 master password.
PWD2	xxxx	Level 2 master password.
PWxx	aaaa . . .	LAPW password number and password.
OVLA	xx xx xx . . .	Overlays accessible by this password.
CUST	xx TEN xxx	Customer number and tenant numbers accessible.
HOST	No	Host mode.
OPT	aaaa . . .	Password options allowed.

LD 22 – Print options for LAPW password (user).

Prompt	Response	Description
REQ	PWD	Print passwords.
PWD2	<CR>	Administrator's password.

Note: Options available to the logged on password are printed. The format is shown below:

PWxx	aaaaaa . . .	LAPW password number and password.
OVLA	xx xx xx . . .	Overlays accessible by this password.
CUST	xx TEN xxx	Customer number and tenant numbers accessible.
Host	No	Host mode.
OPT	aaaa . . .	Password options allowed.

LD 22 – Print contents of Audit Trail buffer (allowed if using PWD1 or PWD2).

Prompt	Response	Description
REQ	PRT	Print.
TYPE	AUDT	Audit Trail.

Feature operation

To bypass a specific restriction imposed by the Limited Access to Overlays feature, enter the appropriate password as defined in LD 17.

Limited Access to Overlays Password Enhancement

Contents

This section contains information on the following topics:

Feature description 739
Operating parameters 740
Feature interactions 740
Feature packaging 740
Feature implementation 740
Feature operation 741

Feature description

The Limited Access to Overlays Password (LAPW) protection mechanism has been enhanced to recognize a LAPW option that can be associated with a user login. Access options are used for ensuring that only a Loss Planning Expert will have the capability to customize entries in any of the Loss Planning tables, including the Static Loss Plan Download (SLPD) table and the Dynamic Loss Switching (DLS) table.

The options provide for Loss Planning data customization Allowed (LOSA) and Loss Planning data customization Denied (LOSD). These password options are configurable using LD 17, and are used to provide password protection for Loss Planning data including the existing Static Loss Plan Download (SLPD) table and the Dynamic Loss Switching (DLS) Alternate Levels table.

Operating parameters

LDs 24 and 88 on all machines have their own passwords. These passwords are unaffected by the Limited Access to Overlay feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Limited Access to Overlays (LAPW) is packaged under package 164.

Feature implementation

LD 17 – The OPT prompt associated with LAPW Password Option Access Rights accepts the access rights for Loss Planning Customization Allowed (LOSA) or Denied (LOSD).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PWD	System password and limited access to overlay password.
- PWD2	xxx...x	Current Level 2 master password.
- LNAME_OPTION	(NO) YES	Require login name for password access?
- NPW1	xxx...x	Level 1 log-in password.
- - LOGIN_NAME	aaa...aaaa	Login name for password access.
- NPW2	xxx...x	Level 2 master password.
- LAPW	0-99	Limited Access password number to change.
- - PWnn	xxx...x	Current LAPW password for password nn.

- - OVLA	xx xx ... xx ALL (XALL)	Overlays (02-99) accessible with PWnn.
- - CUST	0-99 ALL (XALL)	Customers who can access overlays with password PWnn.
- - TEN	xx xx ... xx ALL (XALL)	Tenant list for password access.
- - HOST	(NO) YES	Enable HOST mode log-in for PWnn.
- - OPT	(PSCA PSCD (RDBD) RDBA (LLCD) LLCA (CFPA) CFPD (PROD) PROA (LOSD) LOSA	The following options are accessible with PWnn: Print Speed Call Lists Allowed/Denied Resident Debug Access Denied/Allowed Line Load Control Access Denied/Allowed Change Configuration Allowed/Denied Print Only Access Denied/Allowed, and Loss Plan Customization Denied/Allowed.
- FLTH	0-(3)-7	Failed log-in attempt threshold.
- LOCK	0-(60)-270	Lock-out time in minutes.
- AUDT	(NO) YES	Audit trail (denied) allowed.
- - SIZE	(50)-1000	Word size of audit trail buffer.
- INIT	(YES) NO	Reset locked-out ports on Initialization.

Feature operation

To be able to customize entries in any of the Loss Planning tables, including the Static Loss Plan Download (SLPD) table and the Dynamic Loss Switching (DLS) table, enter the appropriate password as defined in LD 17.

Line and Trunk Cards

Contents

This section contains information on the following topics:

Feature description 743
Operating parameters 745
Feature interactions 746
Feature packaging 746
Feature implementation 746
Feature operation 746

Feature description

In addition to providing a definition for card types, this section lists system cards.

Line Cards

Line Cards provide the interface between the system and telephones, their associated data options, and Attendant Consoles.

- Line Cards
  - NT8D02AA Digital (16 digital telephones plus 16 associated data options)
  - NT8D03AA Analog (16 analog in-line telephones)
- 500/2500 Telephone Line Card

- — QPC594 (4d) (16 ports per card)
- — QPC452 (dd) (eight ports per card)
- — QPC60 (sd) (four ports per card)

- Message Waiting Line Card
  - — NT8D09AA Analog Message Waiting (16 analog single-line telephones with Message Waiting lamps)
  - — QPC789 (4d) (16 ports per card)
  - — QPC494 (dd) (eight ports per card)
  - — QPC267 (sd) (four ports per card)
- SL-1 Telephone Line Card
  - — QPC451 (dd) (eight ports per card)
  - — QPC61 (sd) (four ports per card)
- Attendant Console Line Card
  - — QPC451 (dd) (eight ports per card; four ports per console)
  - — QPC61 (sd) (four ports per card; four ports per console; card must be vintage C or later)
- Integrated Services Digital Line Card (ISDLC)
  - — QPC578 (4d) (16 logical ports per card; eight physical ports; eight for voice/eight for data)

In addition, Data Line Cards are available to interface data communications products.

Note: For information on ITG Line cards, refer to *IP Line: Description, Installation, and Operation* (553-3001-365).

Trunk Cards

Trunk Cards provide the interface between the system and all trunk facilities, including not only public and private network trunks (CO, TIE), but those that connect the system to special features (Recorded Announcement, Paging, and so forth).

- NT8D14AA Universal (any combination of eight: CO, DID, FX, RAN, Paging [low resistance], WATS, TIE, Music)
- NT8D15AA E&M (any combination of four: two-wire E&M, four-wire E&M, four-wire duplex, Paging [high resistance], Emergency Recorder)

Note: For information on ITG Trunk cards, refer to *IP Trunk: Description, Installation, and Operation* (553-3001-363).

Digitone Receivers (DTR)

Digitone Receivers convert Dual-tone Multifrequency (DTMF) signals to a digital format acceptable by the Central Processing Unit (CPU). They are required for all 2500 telephones, some incoming TIE trunks, and Digitone DID trunks. Because DTRs perform a service rather than support an item, the quantity depends on the volume of Digitone traffic generated in a system.

- NT8D16AA Digitone Receiver (eight Digitone Receivers)

Controller Cards

Controller Cards provide the interface and control between the network cards and telephones, consoles, and trunks. These cards are always installed in a dedicated slot in the Intelligent Peripheral Equipment (IPE) module. One Controller Card is required per IPE module.

- NT8D01AD Controller-2 (connects up to two Superloops to one IPE module)
- NT8D01AC Controller-4 (connects up to four Superloops to one IPE module)

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There are no specific packaging requirements associated with this feature.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Line Load Control

Contents

This section contains information on the following topics:

Feature description 747
Operating parameters 749
Feature interactions 750
Feature packaging 750
Feature implementation 750
Feature operation 752

Feature description

Line Load Control (LLC) is a manually activated feature that denies a percentage of call originations from defined groups of stations. Four distinct levels of control are provided:

- LLC OFF Control is set to OFF (default value)
- LLC F Control of First level only
- LLC S Control of Second level only
- LLC T Control of Third level only

When the active Line Load Control (LLC) level is set to OFF, there is no LLC in effect for the system. When the active level is F, S, or T, every line or trunk of the controlled stations has an equal probability of being denied origination. Each LLC level has its own blocking probability percentage (0-100), which is assigned in system software.

The selection of controlled stations is based on the Class of Service of the station or trunk. There are four Class of Service options for LLC:

- LLC N No LLC
- LLC 1 First LLC Class of Service
- LLC 2 Second LLC Class of Service
- LLC 3 Third LLC Class of Service

The control levels are enabled manually through LD entry and operate in a hierarchical manner. Only one control level can be active at a time. Progressive in sequence, each operating level restricts another class of stations and the classes below it.

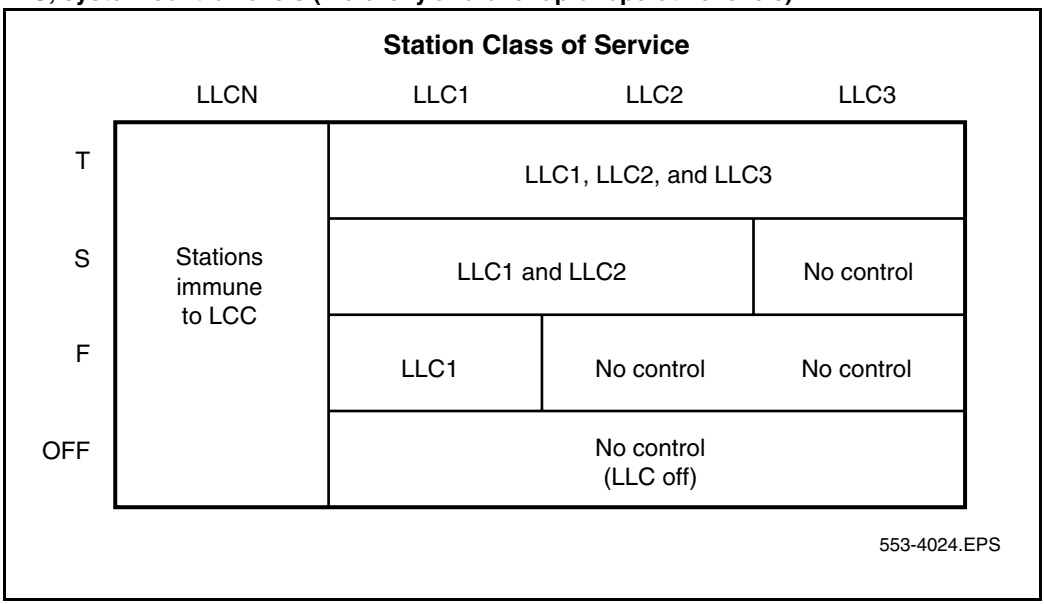
Figure 18 describes the hierarchical nature of LLC. Restrictions are based on the number of originating calls blocked by the probability level set in the LD program.

For example, when LLC S level is enabled, all stations with LLC 1 and LLC 2 Class of Service are limited by the feature, while LLC 3 calls function normally. When LLC T is enabled, only those stations with LLC N Class of Service are allowed to originate calls without restrictions.

Probability levels set by the LD program are whole numbers between 0 and 100. A probability set at 0 (the default value) means no call origins are restricted for that Class of Service. A probability setting of 100 means all calls are restricted when that Class of Service is enabled. Numbers between 0 and 100 are treated as a percentile of calls blocked.

During call processing, LLC screens calls to find the Class of Service for that Directory Number (DN) and the active LLC level, and then decides if the originating set is to receive a dial tone. Sets that are blocked during an LLC level upgrade do not receive a dial tone.

Figure 18
LLC, system control levels (hierarchy and overlap of operative levels)



Operating parameters

The following services are not subject to LLC:

- Attendant stations
- Direct Inward System Access (DISA), and
- Hot Line services.

Established calls are not affected by LLC upgrades, only new calls attempted.

The system counts the calls denied for each Class of Service, and prints the traffic data periodically as part of the Processor Load Format TFS004.

Feature interactions

Automatic Redial

Automatic Redial (ARDL) attempts are controlled and restricted by Line Load Control.

Feature packaging

Line Load Control (LLC) package 105 must be enabled for this feature to operate.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Add or change Line Load Control for analog (500/2500 type) telephones.

2 LD 11 – Add or change Line Load Control for Meridian 1 proprietary telephones.

3 LD 2 – Set Line Load Control levels.

LD 10 – Add or change Line Load Control for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(LLCN) LLC1 LLC2 LLC3	LLC not enabled (the default). LLC class 1. LLC class 2. LLC class 3.

LD 11 – Add or change Line Load Control for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(LLCN) LLC1 LLC2 LLC3	LLC not enabled (the default). LLC class 1. LLC class 2. LLC class 3.

LD 2 – Set Line Load Control levels.

Prompt	Response	Description
SCTL	x aaa	Set blocking probability. x = F (LLC, level F). S (LLC, level S). T (LLC, level T). aaa = 0-100.

SLLC	x	Activate LLC at level x. x = F (LLC, level F). S (LLC, level S). T (LLC, level T). OFF (deactivate LLC)
TLLC		Print blocking probability and current active LLC level.

Feature operation

No specific operating procedures are required to use this feature.

Line Lockout

Contents

This section contains information on the following topics:

Feature description 753
Operating parameters 754
Feature interactions 754
Feature packaging 756
Feature implementation 756
Feature operation 757

Feature description

When a user remains off hook without dialing any digits, a timeout occurs. The transmission path is released for other uses. Dial tone timeout and interdigit timeout for telephone and Direct Inward System Access (DISA) trunks are considered Line Lockout situations.

The 2500 telephones lock out after 15 seconds. Meridian 1 proprietary telephones, and 500 telephones lock out after 30 seconds. When Line Lockout occurs, the system gives overflow tone for 14 seconds and then puts the telephone in a lockout state. Meridian 1 proprietary telephones are idled, and analog (500/2500 type) telephones appear busy to any incoming calls. DISA calls receive overflow tone.

Flexible Line Lockout-This feature provides three options for lockout treatment for stations and DISA calls. Flexible Line Lockout can perform any of the following functions:

- provide the existing overflow tone and then lockout treatment
- immediately intercept calls to the attendant, or
- receive overflow tone and then intercept to the attendant.

When a call is intercepted to the attendant, ringback is returned and the call appears at the Attendant Console on a designated Line Lockout (LCT) Incoming Call Indicator (ICI) key. If an LCT ICI key is not defined, the call is treated as a normal incoming call.

When the attendant answers the call, the Directory Number (DN) of the originating telephone, followed by the name (if Call Party Name Display is enabled), is displayed on the console. The attendant may then terminate the call or offer assistance to the call originator.

Flexible Line Lockout Timers – This enhancement to Flexible Line Lockout provides three variable Line Lockout timers. The timers are defined in LD 15, and range from 0 to 60 seconds.

Operating parameters

TIE trunk calls do not receive overflow tone during Line Lockout, and do not receive Flexible Line Lockout treatment.

Feature interactions

Attendant Blocking of Directory Number

If an Attendant Blocking of DN attempt is made on a set in Line Lockout state, busy tone will be returned.

Attendant Overflow Position

A call intercepted to the attendant due to Flexible Line Lockout receives Attendant Overflow Position (AOP) treatment if the feature package is equipped and the AOP Directory Number (DN) is defined.

Call Detail Recording

If a Direct Inward System Access (DISA) call routes to the attendant due to Flexible Line Lockout, and Call Detail Recording (CDR) is selected for incoming trunk calls, a call record generates when the attendant terminates the call after answer. The CDR record shows the attendant number and the route and member numbers.

If the attendant extends the call, the CDR record generates when the call is terminated. The CDR record does not show the attendant Directory Number (DN).

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is not allowed for any telephone that is in Line Lockout state.

Direct Inward System Access

The defined Flexible Line Lockout treatment is provided to Direct Inward System Access calls.

Display

If a call from a telephone equipped with a display is intercepted to the attendant due to Flexible Line Lockout, the telephone displays the digits dialed, if any, before the intercept. If no digits are dialed, the attendant DN and name (if configured) will be displayed. When the attendant answers the call, the console displays the DN and the number zero (0), or any digits dialed and the name (if configured) of the telephone intercepted.

Off-Hook Alarm Security

Off-Hook Alarm Security treatment occurs when a telephone with ASCA Class of Service receives an interdigit or dial tone timeout. The ASTM is used instead of the dial tone and interdigit timers (DIDT and DIND, respectively) normally used for LLT and DLT line lockout treatment.

Recorded Overflow Announcement

Calls intercepted to the attendant due to Flexible Line Lockout receive Recorded Overflow Announcement (ROA) treatment if the Line Lockout (LCT) Incoming Call Indicator (ICI) key is configured for ROA.

System Overflow Tone

If the option for Flexible Line Lockout to the attendant is enabled, any call that is given overflow tone (for example, if the wrong access code is dialed, or if the telephone is not allowed to dial the Trunk Access code) is intercepted to the attendant on overflow timeout.

Feature packaging

This feature is included in base system software.

Feature implementation

LD 15 – Implement Flexible Line Lockout for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	INT	Intercept treatment options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
ICI	0-19 LCT	Assign a Flexible Line Lockout Incoming Call Indicator (ICI) key to Attendant Consoles.
- LLT		Line Lockout treatment.
	(OVF) OFA ATN	Overflow tone, then lockout. Overflow tone, then attendant intercept. Attendant intercept.

- DLT		Line lockout treatment for Direct Inward System Access (DISA) calls.
	(OVF)	Overflow tone, then lockout.
	OFA	Overflow tone, then attendant intercept.
	ATN	Attendant intercept.
TYPE	TIM	Timers.
- DIND	2-(30)-60	Dial tone and interdigit timeout for Meridian 1 proprietary telephones, and 500 telephones.
- DIDT	2-(14)-60	Dial tone and interdigit timeout for 2500 telephones.
- BOTO	2-(14)-60	Busy tone and overflow tone timeout for all telephones.

Feature operation

No specific operating procedures are required to use this feature.

Listed Directory Numbers

Contents

This section contains information on the following topics:

Feature description 759
Operating parameters 759
Feature interactions 760
Feature packaging 760
Feature implementation 760
Feature operation 761

Feature description

Each customer within the system can have up to four Listed Directory Numbers (LDNs) in the public directory on Direct Inward Dialing (DID) trunks. Each Listed Directory Number (LDN) is assigned to an Incoming Call Indicator (ICI) key, enabling the attendant to answer an incoming call appropriately. For systems without DID facilities, LDNs can be provided on incoming Public Exchange/Central Office (CO) trunks assigned to a trunk group and an Incoming Call Indicator (ICI) key on the console. Local telephones and TIE trunks can call the attendant using any of the four DNs.

Operating parameters

A maximum of four LDNs can be assigned per customer.

Feature interactions

Call Forward No Answer

A Listed Directory Number (LDN) that is assigned to an Incoming Call Indicator (ICI) has a higher priority than a Call Forward No Answer ICI. When a call is forwarded to an LDN via Flexible DN, the call is presented on an LDN ICI.

Call Party Name Display

Call Party Name Display (CPDN) is not supported for LDNs. If the LDN call is from an incoming trunk route, the CPND assigned to the route access code is displayed.

Directory Number Expansion

LDNs can have up to seven digits if the Directory Number Expansion (DNXP) package is equipped.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Assign Listed Directory Numbers for each customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	LDN	Departmental Listed Directory Numbers
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- LDN0	xxx...x	LDN0.
- LDN1	xxx...x	LDN1.

- LDN2	xxx...x	LDN2.
- LDN3	xxx...x	LDN3.

Feature operation

No specific operating procedures are required to use this feature.

Listed Directory Numbers, Network Wide

Contents

This section contains information on the following topics:

Feature description 763
Operating parameters 764
Feature interactions 764
Feature packaging 765
Feature implementation 766
Feature operation 768

Feature description

Listed Directory Numbers (LDNs) can be defined as Incoming Call Indicator (ICI) keys on an Attendant Console, making it possible to have different presentations when different DNs are dialed. This feature makes it possible to define six LDNs on a system.

If the dialed DN is an LDN and an LDN key exists corresponding to the dialed LDN, the call will be presented on that ICI LDN key.

This feature also enables LDNs to be recognized network wide when Network Attendant Service (NAS) is used. The same LDNs must be configured in multiple nodes. Network LDN is defined on a customer basis.

Operating parameters

The network part of this feature works in a Meridian Customer Defined Network (MCDN) environment with NAS configured.

The LDNs to be used network wide cannot be used in conjunction with Distant Steering Codes.

Feature interactions

Call Forward No Answer

With this feature, the LDN ICI has a higher priority than CFNA ICI. When a call is forwarded to an LDN via Flexible DN, the call will be presented on the LDN ICI.

Centralized Attendant Service

Centralized Attendant Service (CAS) is mutually exclusive to the NAS package. As the network wide LDN feature requires NAS for its networking functions, the network part of this feature will not work with CAS, but the two extra LDNs can be used locally.

Console Operation - Console Presentation

Console Operation makes it possible for each console to select which ICI call types will be presented to the console. Network wide LDN does not work with the Console Presentation feature because it is not supported by NAS. Console Operation can, however, be configured with two additional LDNs.

Console Operation - Queue Thermometer

The queue thermometer indicates how many calls are in the queue for a certain ICI key. An ICI key can correspond to more than one ICI type. Even though the ICI type of a call may be different with or without this feature active, it will not interact with queue thermometer operations.

Console Presentation Group Level Services

This feature provides two more LDNs per Console Presentation Group.

Departmental Listed Directory Number

Departmental LDN is not supported over the network; however, this feature does provide two more LDNs for the DLDN feature.

Network Attendant Service

The way the network LDN calls are presented in a NAS environment is changed by this feature. The presentation on the NDID, NTIE, NCO, NFEX or NWAT, and the LDN0 key is changed to the correct LDN key, if it exists. Otherwise, it will be presented as it previously was on the NDID or LDN0 key.

Network Message Center

With this feature, the LDN ICI has a higher priority than MWC ICI. When a call is forwarded to an LDN over the network to a message center, the call will be presented on the LDN ICI.

Feature packaging

Since Network Wide LDN requires Network Attendant Service routing, the following existing software packages must be provisioned: Network Attendant Service (NAS) package 159; Network Alternate Route Selection (NARS) package 58; Network Class of Service (NCOS) package 32; Basic Routing (BRTE) package 14; and applicable ISDN options depending upon customer requirements.

To use the attendant queue thermometer, Console Operations (COOP) package 169 must be provisioned.

For Departmental LDN to be configured with six LDNS, Departmental LDN (DLDN) package 76 must be provisioned.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Activate Network Wide LDN in CDB. Four prompts define the extended LDN numbers and the Listed Attendants (LDAs) belonging to the LDNs. The prompts can be answered in the same way as the prompts LDN0, 1, 2, 3. The LDA prompts only appear if DLDN is set to YES. These store the Attendant Console number associated with the LDN number.

2 LD 15 – Add or change LDN keys.

3 LD 93 – Add or change LDN keys in CPG.

LD 15 – Activate Network Wide LDN in CDB. Four prompts define the extended LDN numbers and the Listed Attendants (LDAs) belonging to the LDNs. The prompts can be answered in the same way as the prompts LDN0, 1, 2, 3. The LDA prompts only appear if DLDN is set to YES. These store the Attendant Console number associated with the LDN number.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	CDB LDN	Customer Data Block. Departmental Listed Directory Numbers.
...		
- DLDN	YES	YES if no Console Presentation Group (CPG) is configured.
...		
- LDN4	xxxx(xxx)	Listed Directory Number 4. If the DN Expansion (DNXP) package is equipped, up to seven digits are allowed; otherwise, only four digits are allowed.
- LDA4	xx xx... ALL	Attendant Consoles associated with LDN4.

- LDN5	xxxx(xxx)	Listed Directory Number 5. If the DNXP package is equipped, up to seven digits are allowed; otherwise, only four digits are allowed.
- LDA5	xx xx... ALL	Attendant Consoles associated with LDN5.
- OPT	NLDN (XLDN)	Enable network wide LDN. Exclude LDN.

LD 15 – Add or change LDN keys.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	LDN	Departmental Listed Directory Numbers.
...		
- ICI	x LD4	Listed DN 4, where x is the key number.
- ICI	x LD5	Listed DN 5, where x is the key number.

LD 93 – Add or change LDN keys in CPG.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	CPGP	Changes affect the Console Presentation Group parameters.
CUST	xx	Customer number, as defined in LD 15
CPG	x	CPG number.
...		
LDN4	xxxx(xxx)	Listed Directory Number 4. If the DNXP package is equipped, up to seven digits are allowed; otherwise only four can be entered.

LDN5	xxxx(xxx)	Listed Directory Number 5. If the DNXP package is equipped, up to seven digits are allowed; otherwise only four can be entered.
...		
ICI	x LD4	x is the key number for listed DN 4.
ICI	x LD5	x is the key number for listed DN 5.

Feature operation

Calls to node 1 on an LDN, routed by NAS to node 2, are presented to the attendant on node 2 on an ICI according to the following rules.

Note: The feature option in the origination and terminating node is turned on.

1 If an LDN key exists corresponding to the dialed DN, the call is presented on this LDN ICI key.

2 If no LDN key corresponding to the dialed DN exists, and an ICI key for the trunk type exists, the call is presented on a matching trunk type key.

3 If neither of the above cases exists, the call is presented to LDN key 0.

4 If there is no LDN zero and no trunk type ICI keys, the call is only presented on the loop key.

Lockout, DID Second Degree Busy, and MFE Signaling Treatments

Contents

This section contains information on the following topics:

Feature description 769
Operating parameters 770
Feature interactions 770
Feature packaging 771
Feature implementation 771
Feature operation 771

Feature description

This feature allows networking treatment to be applied to Multifrequency Signaling for Socotel (MFE), provides an intercept treatment for sets in lockout state, and allows calls to Second Degree Busy sets to be disconnected or routed to the attendant.

These components are described below:

- Calls to a telephone set in lockout state are given full intercept treatment, rather than receiving busy tone. Depending on the configuration, the calls are either routed to the attendant, or given overflow tone. This treatment applies to standalone and networking environments.

- Direct Inward Dialing (DID) calls to a telephone set in Second Degree Busy (that is, a set that is busy on a call, and has another call waiting or camped-on) state are either disconnected, receive busy tone, or routed to the attendant. If the Second Degree Busy Disconnect (DSTD) option is defined, the call treatment depends on the Class of Service of the second degree busy telephone set; Forward Busy Allowed (FBA) causes the calls to be call forwarded busy to the attendant, while Forward Busy Denied (FBD) causes the calls to receive busy tone.
- MFE signaling provides call status information for DID calls over MFE-registered trunks. If a call tandems across an Integrated Services Digital Network (ISDN) network, this enhancement allows the call status information to be sent to the incoming MFE trunk from any outgoing ISDN trunk.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Automatic Call Distribution

The lockout and second degree busy treatments do not apply to Automatic Call Distribution DNs.

Call Forward
Call Forward Busy
Hunting
Message Waiting Forward Busy
Flexible Feature Code (FFC) Boss Secretarial Filtering

Call Forward, Call Forward Busy, Call Hunt, Message Waiting Forward Busy, and FFC Boss Secretarial Filtering take precedence over lockout and second degree busy.

Feature packaging

This feature is packaged as International Supplementary Features (SUPP) package 131; Network Attendant Service (NAS) package 159; Integrated Services Digital Network (ISDN) package 145; and Multifrequency Signaling (MFE) package 135.

Feature implementation

LD 15 – Define an option for DID calls to a second degree busy telephone, and define a new intercept treatment for calls in a lockout state.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	FTR	Features and options.
...		
- OPT	(DSTD) DSTA	DID call to Second Degree Busy treatment (denied) allowed. If allowed, DID calls forwarded to a busy set are disconnected. If denied, calls forwarded to a busy set follow the set's CLS (FBA/FBD) treatment.
TYPE	INT	Intercept treatment options.
...		
INTR	YES	Change Intercept Treatment.
- LCKT	(BSY) OVF ATN RAN NAP SRC1 SRC8	Four of these entries must be entered. The default value is BSY BSY BSY BSY.

Feature operation

No specific operating procedures are required to use this feature.

LOGIVOX Telephone

Contents

This section contains information on the following topics:

Feature description 773
Operating parameters 773
Feature interactions 774
Feature packaging 774
Feature implementation 774
Feature operation 775

Feature description

The LOGIVOX is a Swedish telephone similar to the SL-1 telephone, but designed to work on the Swedish A345 500/2500 (a system with modified software). The system echoes dialed digits to the telephone, while the A345 does not. The LOGIVOX uses its own firmware to display dialed digits. Therefore, to allow the use of the LOGIVOX telephone with the system, a Class of Service is provided that suppresses dialed digits from the system, including Last Number Redial. All other digit-display messages are provided through the system, as required. Expanded LOGIVOX telephones, with up to two extra key/lamp strips also may be configured, as required.

Operating parameters

Call party name display is not supported on LOGIVOX telephones.

The LVXA Class of Service cannot be defined or changed through Attendant Console LD 12. In addition, LXVA Class of Service telephones cannot be tested through LD 31.

A telephone assigned LXVA Class of Service cannot be a maintenance set.

The LVXA Class of Service should only be given to a LOGIVOX telephone.

Feature interactions

Digit Display

During manual dialing or last number redial, the display shows the dialed digits, even if the set has display denied Class of Service. If the set has LOGIVOX denied Class of Service, each digit is shown twice.

On-hook Dialing

Because of the firmware on the LOGIVOX set, the DN key 0 is automatically selected when the first digit is dialed, and no other DN has been selected.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

LD 11 – Modify the system hardware and software parameters to allow logivox Class of Service.

Prompt	Response	Description
...		
CLS		Class of Service.
	(NDD) ADD DDS	No Digit Display. Automatic Digit Display. Digit Display Standard.
	(LVXD) LVXA	LOGIVOX Class of Service (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Loop Start Supervisory Trunks

Contents

This section contains information on the following topics:

Feature description 777
Operating parameters 778
Feature interactions 778
Feature packaging 779
Feature implementation 779
Feature operation 780

Feature description

This feature permits the system to detect disconnect and answer supervision, when provided by the Public Switched Telephone Network (PSTN), for outgoing Central Office (CO), FEX, or WATS loop-start trunks. Answer and disconnect supervision signals, provided by the PSTN and subsequently detected by the system, reverse the battery polarity on the tip and ring leads of the trunk (reverse-battery signaling).

Polarity Sensitive Packs (PSPs) or Polarity Insensitive Packs (PIPs) are identified in LD 14.

This feature has the following options:

Toll Definition Coincident

The toll definition allows any digit dialed as the first digit after the trunk access code to define the call as a toll call (refer to LD 16).

Answer Supervision

An answer supervision signal received from the PSTN indicates the call is established for the purpose of other features such as Call Detail Recording (CDR) with answer supervision.

Disconnect Supervision

A disconnect supervision signal is sent when either the calling or called party disconnects thereby freeing the trunk for other use.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Automatic Call Distribution

Since Loop Start Supervisory trunks do not provide disconnect supervision on incoming calls it is not recommended that these trunks be used to auto terminate on an Automatic Call Distribution (ACD) DN.

Call Detail Recording

Call Detail Recording (CDR) will use the toll definition digits as defined in a trunk's Route Data Block instead of using "0" or "1" to identify toll calls.

Call Detail Recording with Answer Supervision

For outgoing calls, the Answer Supervision received from the far end, on Loop Start trunks, will determine when the "CDR with Answer Supervision" feature will start recording the duration of the call.

Call Transfer

If an internal station user transfers an answered outgoing call to another station in the ringing state, then any disconnect signal received from the far end causes the trunk to be released and ringing of the internal set to stop. This operation eliminates the problem of holding trunks and extensions due to lack of supervision on Loop Start trunks.

China – Busy Tone Detection

The interaction with Intelligent Peripheral Equipment (IPE) trunks occurs because Busy Tone Supervision (BTS) can be configured in conjunction with any existing supervision type. For the EXUT, BTS can now be configured with a supervision type of BST (both incoming and outgoing battery reversal) and Polarity Insensitive (PIP). These supervision type's call processing methods are not changed, except that now the first type of supervision received is the one acted upon.

1.5 Mbit Digital Trunk Interface

The CO Loop Start Supervisory trunk will not be supported as a 1.5 Mbit Digital Trunk Interface (DTI) type.

Feature packaging

Loop Start Supervisory Trunks is included in base system software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Create or modify trunk data blocks on a per trunk basis.

2 LD 16 – Create or modify trunk route data blocks.

LD 14 – Create or modify trunk data blocks on a per trunk basis.

Prompt	Response	Description
...		
SIGL	LOP	Loop start supervision.
SUPN	YES (NO)	Trunk Supervision required (not required).
STYP	PSP (PIP)	Polarity sensitive packs. Polarity insensitive packs.

LD 16 – Create or modify trunk route data blocks.

Prompt	Response	Description
...		
NATL	(YES) NO	North American toll scheme (a toll call has 0 or 1 as first digit after the trunk access code). Prompted when SUPP package is equipped or OAL = YES or OTL = YES.
TDG	0-9	Toll digits – list of digits after the trunk access code which indicates toll calls. Prompted when NATL = NO.

Feature operation

No specific operating procedures are required to use this feature.

Loop Start Supervisory Trunks (Incoming Calls)

Contents

This section contains information on the following topics:

Feature description 781
Operating parameters 782
Feature interactions 782
Feature packaging 782
Feature implementation 783
Feature operation 783

Feature description

This feature adds disconnect supervision for incoming calls from the Public Switched Telephone Network (PSTN) or Central Office (CO), FEX, or WATS loop-start trunks. This is in addition to the existing answer and disconnect supervision available on outgoing trunks for the loop start supervisory trunk feature.

The disconnect supervision on incoming calls applies only to Polarity Insensitive Packs (PIPs). It is the change in polarity (reverse battery), rather than the absolute polarity, that must be detected.

A change in polarity from the PSTN side indicates that the calling party has discontinued the call. The detection of this supervision signal allows a Call Detail Recording (CDR) record to be produced and the trunk to be idled.

Operating parameters

The Central Office cannot disconnect until one second after it is answered by an attendant or station.

This feature is not compatible with the Japan Trunk feature, on a trunk basis.

Polarity detection is disabled during outpulsing. Therefore, polarity state changes of less than 200 milliseconds are ignored after trunk seizure, as are power interruptions of unlimited duration.

If a system station goes on-hook first, a far-end disconnection cannot be detected.

Feature interactions

Automatic Call Distribution

Loop Start trunks with Both Way Supervisory (BST) Class of Service may be used to auto terminate on an Automatic Call Distribution (ACD) DN. Caller disconnection can be detected on trunks designated as BST and removed from the ACD queue.

Call Modification

If an incoming call that is transferred by the attendant to a station is in the ringing state, and the far-end (the Central Office) disconnects, the trunk is released and the ringing stops.

Integrated Voice Messaging Service and Integrated Messaging Service

Integrated Voice Messaging Service (IVMS) and Integrated Messaging Service (IMS) use ACD queues, therefore trunks designated BST may the used for these services.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 14 – Create or modify trunk data blocks on a per trunk basis.

Prompt	Response	Description
SIGL	LOP	Loop start supervision.
SUPN	YES (NO)	Trunk Supervision required (not required).
STYP	BST	Both way Supervisory Trunk - Supervision on both incoming and outgoing loop start PSTN (CO) trunks.

Feature operation

No specific operating procedures are required to use this feature.

Loopback on Public Exchange/ Central Office Trunks

Contents

This section contains information on the following topics:

Feature description 785
Operating parameters 785
Feature interactions 786
Feature packaging 786
Feature implementation 786
Feature operation 786

Feature description

When a Loop Start signaling arrangement Public Exchange/Central Office (CO) trunk unit is disabled a loopback is performed – the unit is hardware seized to prevent the far end switch from making an incoming call; the CO trunk appears to be in an off-hook state. This enhancement prevents loopback from being performed in this scenario.

Operating parameters

This enhancement applies to the Central Office trunk card used in France, which is the NTD9742A.

This enhancement does not apply to CO trunk cards located on Intelligent Peripheral Equipment shelves (loopback prevention is handled by the trunk card in this configuration).

This enhancement does not apply to Direct Inward Dialing (DID)/Direct Outward Dialing (DOD) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires French Type Approval (FRTA) package 197.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

M2312 Digit Display

Contents

This section contains information on the following topics:

Feature description 787
Operating parameters 787
Feature interactions 788
Feature packaging 788
Feature implementation 788
Feature operation 789

Feature description

This feature supports the addition of a two line by 24-character digit display to the M2112 telephone, making it possible to assign an M2112 digital telephone one of the Display Allowed (ADD or DDS) Classes of Service. With either of these classes assigned, the M2112 digital telephone with digit display (M2312) will display digits in a manner similar to the SL-1 telephone.

The display format is the same as that used for the SL-1 telephone, except that 24 characters are available (instead of 16).

Operating parameters

The M2312 telephones will operate on either double density or quadruple density loops. Those telephones configured on a double density loop will be capable of voice service only. Those telephones that are configured on quadruple density loops can provide integrated voice and data services.

Feature interactions

Call Party Name Display

The calling party number can be displayed only when the call is active.

Hold

The digit display will go blank when a call is placed on hold.

Mute

Muting a call will not affect the digit display.

SL-1 digit display

The first line of the digit display can show the characters 0-9, *, #, P, and - The M2312 digit display differs in this respect from that of the SL-1 telephone; the M2312 can display * and #, and hence does not use a space or the H character to represent them.

Feature packaging

M2000 Digital Sets (DSET) package 88.

Feature implementation

LD 11 – Create or modify the data blocks for Meridian 1 proprietary telephones.

Prompt	Response	Comment
TYPE:	xxxx	Digital data block for xxxx digital set.
CLS		Class of Service.
	(NDD) ADD DDS	No Digit Display, Automatic Digit Display, or Digit Display Standard.

Feature operation

The first line will be capable of displaying the same characters as the SL-1 telephone's digit display. The second line will display the date and time. In addition, when a call is active on key 0, a call timer will be displayed on the second line.

The following display lines can be called up by manual key operations:

- date and time
- buzz DN
- call waiting party
- voice call party
- autodial number
- speed call number
- ring again party
- call forward party
- call pickup

The following display lines can be automatically displayed:

- dialed number, and
- number of calling party.

The time and date function shown on the second display line is generated within the telephone. However, the telephone clock is automatically updated at least once a day from the switch's system clock. The call timer that appears on the second line is generated and controlled completely within the telephone. The function is not controlled by the switch.

Make Set Busy and Voice Call Override

Contents

This section contains information on the following topics:

Feature description 791
Operating parameters 791
Feature interactions 792
Feature packaging 792
Feature implementation 792
Feature operation 793

Feature description

This feature allows an incoming voice call to override the Make Set Busy feature activated on a Meridian 1 proprietary telephone, and to terminate on the set. The set is given a two-second burst of ringing tone before the call connection is established.

All other incoming call types remain blocked by Make Set Busy.

Operating parameters

A Voice Call key on a Meridian 1 proprietary telephone can only be programmed to a single appearance DN.

The set being voice called must be equipped with a speaker.

Feature interactions

Do Not Disturb

Voice calls are not allowed on a set with attendant-activated Do Not Disturb.

Make Set Busy

This feature allows an incoming voice call to override the Make Set Busy feature activated on a Meridian 1 proprietary telephone, and to terminate on the telephone. The telephone is given a two-second burst of ringing tone before the call connection is established.

All other incoming call types remain blocked by Make Set Busy.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Enable Make Set Busy Voice Call Override.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- OPT	VOBA	Voice Override Busy allowed. The response to the OPT prompt has to be VOBA to allow a voice call to override a Make Set Busy condition.

Feature operation

The following example illustrates how a voice call can be made to a set with MSB active:

In this example, Set A is a Meridian 1 proprietary telephone with a **VCC** key programmed with the DN of a single appearance key on set B.

Set B is a Meridian 1 proprietary telephone with a single appearance DN key. Set B has a **Make Set Busy** key which has been activated.

1 A goes off-hook, and receives dial tone.

2 A presses the **VVC** (Voice Call) key.
A's VCC key lamp is lit and A receives ringback tone. B receives a two-second burst of ring tone. B's terminating DN key lamp flashes.

3 After two seconds:
Set A has a one-way voice path to set B. B's DN key lamp is lit. Ring tone to B stops. Ringback tone to A stops. B's Make Set Busy lamp remains lit.

4 If B goes off-hook, A and B are connected in a normal two-way conversation.

Make Set Busy

Contents

This section contains information on the following topics:

Feature description 795
Operating parameters 796
Feature interactions 796
Feature packaging 800
Feature implementation 800
Feature operation 801

Feature description

The Make Set Busy (MSB) feature allows a Meridian 1 proprietary telephone to appear busy to all incoming calls. Outgoing calls can still be made from the telephone. To activate this feature, a separate MSB key/lamp pair must be assigned. Incoming calls to Multiple Appearance Directory Numbers (MADNs) in the MSB mode are still signified by the indicator next to the Directory Number (DN) key, and can be answered even while MSB is active. Calls to any Single Appearance Directory Number on the telephone receive a busy indication.

Make Set Busy Flexible Feature Codes

You can activate Make Set Busy from an analog (500/2500 type) telephone by dialing the Make Set Busy Activate (MSBA) FFC (defined in LD 57). To deactivate Make Set Busy, the user dials the Make Set Busy Deactivate (MSBD) FFC (defined in LD 57) or the general Deactivate (DEAF) FFC (also defined in LD 57).

Operating parameters

Make Set Busy does not affect incoming Private Line calls.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will override the Make Set Busy feature. If the dialed DN of the set that has the Make Set Busy feature is idle, the DN will be blocked and if the DN is busy, busy tone will be heard.

Attendant Break-In

For a telephone with Make Set Busy in effect, Break-In is temporarily denied to the attendant. The Break-In lamp uses a slow flash to indicate this situation. Using the Break-In key prior to dialing the destination DN circumvents this situation. After the Break-In, the telephone returns to its prior status.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Make Set Busy that may be applied to the set.

Attendant Overflow Position

If a telephone that is the only idle AOP DN has MSB activated, calls will not overflow.

If the AOP DN is a multiple appearance DN, the MSB key should be added to all telephones with an AOP DN.

If MSB is activated in a Multiple Call Ringing arrangement, the telephone appears busy. All other appearances of the AOP DN will still receive calls. This allows the user to leave the telephone and prevent callers from overflowing and receiving ringback with no answer.

If the AOP DN is a Multiple Appearance, Single Call arrangement and MSB is activated, the AOP DN of that telephone will flash, but the telephone will not ring (the call can still be answered from that appearance).

Automatic Call Distribution

See *Automatic Call Distribution: Description* (553-3001-351) for information on MSB operations.

Automatic Set Relocation

If Make Set Busy is active when the telephone is relocated, Make Set Busy remains active.

Busy Lamp Field

When a Make Set Busy key is activated, the Busy Lamp Field array will indicate that the first DN only on that set is busy.

Call Forward All Calls

Call Forward All Calls and then Hunting take precedence over MSB.

Call Forward/Hunt Override Via Flexible Feature Code

Make Set Busy is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Call Park

Recall of a parked call to a telephone in the Make Set Busy mode is intercepted by the attendant.

Camp-On, Forced

Telephones with Make Set Busy active cannot be camped on to with Forced Camp-On. Overflow tone is returned to telephones attempting Forced Camp-On. Voice Call is blocked by Make Set Busy.

China – Attendant Monitor

If an attendant attempts to monitor a DN which has Make Set Busy activated and is idle, idle DN treatment is given.

China – Flexible Feature Codes - Customer Call Forward

Enhanced Flexible Feature Codes - Customer Call Forward

Customer Call Forward takes precedence over Make Set Busy if both are active.

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

Executive Intrusion is not allowed if either of these features is active at the requested party.

Flexible Feature Code enhancement

The Deactivate FFC can be used to deactivate Make Set Busy.

Group Call

A Group Call to a telephone in Make Set Busy or Individual Do Not Disturb mode cannot be completed. The telephone will not be rung and is not counted as part of the Group Call (that is, if all other members in the group have answered, the lamp next to the Group Call key on the originator's telephone lights steadily).

Group Hunt

Make Set Busy (MSB) has priority over Group Hunting. Group Hunting will skip over sets with MSB active.

Hot Line

Make Set Busy is overridden by the Hot Line feature. If a Meridian 1 proprietary telephone is in Make Set Busy mode, incoming Hot Line calls still terminate (ring) on the telephone.

> ***Note:*** The Conference-Hot Line key overrides Make Set Busy only when the terminating key is HOT.

Idle Extension Notification

It is not possible to request Idle Extension Notification towards an extension that has the Make Set Busy feature activated.

If Idle Extension Notification is requested for a Multiple Call Arrangement DN, the first extension with this DN that becomes idle will cause the recall. This extension will also be blocked from receiving calls.

ISDN QSIG/EuroISDN Call Completion

Sets that have Make Set Busy (MSB) activated can request Call Completion to another DN, as the free notification overrides the MSB feature. Incoming Call Completion to Busy Subscriber (CCBS) requests do not override the MSB feature. A set is considered busy while MSB is active. A CCBS request is registered against a busy set, but only advances when the MSB feature is deactivated and the set remains free.

Make Set Busy and Voice Call Override

This feature allows an incoming voice call to override the Make Set Busy feature activated on a Meridian 1 proprietary telephone, and to terminate on the telephone. The telephone is given a two-second burst of ringing tone before the call connection is established.

All other incoming call types remain blocked by Make Set Busy.

Network Individual Do Not Disturb

The Individual Do Not Disturb (DNDI) intercept treatment takes precedence over Make Set Busy indication.

Network Intercom

Hot Type I calls terminating on a station in the Make Set Busy mode override Make Set Busy.

Override

Telephones with MSB active cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.Voice Call is blocked by MSB.

Override, Enhanced

Priority Override

Telephones with MSB active cannot be affected by Priority Override. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Feature packaging

Make Set Busy (MSB) package 17 has no feature package dependencies.

The following packages are required for Make Set Busy FFCs:

- Background Terminal Facility (BGD) package 99.
- Flexible Feature Codes (FFC) package number 139, and

Feature implementation

LD 11 – Add or change MSB for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx MSB	Add an MSB key (must be key 30 for M3000 telephones). xx = key number.

Feature operation

To make a telephone appear busy to callers:

- Without lifting the handset, press the MSB key. The indicator lights steadily and the telephone will not receive calls.

To cancel MSB:

- Without lifting the handset, press the MSB key. The indicator light is extinguished.

The following instructions are for using Make Set Busy FFCs:

- Activate
  The user must dial the Make Set Busy Activate (MSBA) FFC.
- Deactivate
  The user must dial the Make Set Busy Deactivate (MSBD) FFC or the Deactivate (DEAF) FFC.

Make Set Busy Improvement

Contents

This section contains information on the following topics:

Feature description 803
Operating parameters 804
Feature interactions 804
Feature packaging 805
Feature implementation 806
Feature operation 807

Feature description

This feature is designed for a boss/secretary environment. The same Directory Number (DN) appears on more than one set, and is defined as ringing on the secretary set and non-ringing on the boss set.

The Make Set Busy Improvement (MSBI) feature provides an audible notification to the executive non-ringing DN, when all of the secretaries have activated the Make Set Busy (MSB) key on the same appearance of the DN.

Example: The incoming call is directed to the executive DN, the key lamp flashes on the executive set, the secretary receives an audible notification of the same call. If the secretary is not available to answer the call, the secretary presses the MSB key and the call goes back to the executive with audible notification (buzzing or ringing).

The MSBI feature is configured as a new Class Of Service, Make Set Busy Improvement Allowed (MSIA) or Make Set Busy Improvement Denied (MSID). The MSBI feature is configured on a specific Terminal Number (TN) and affects the Single Call Non-Ringing (SCN), Multiple Call Non-Ringing (MCN) and the Private Line Non-Ringing (PVN) keys on that specific TN.

Operating parameters

This feature can be used on proprietary sets with DN key type SCN, MCN or PVN.

The MSBI feature does not support data terminals, Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) sets or analog (500/2500) sets. However, the ringing appearances of the DN can be a (500/2500) set but not for a private line.

Feature interactions

Directory Number Delayed Ringing

The MSBI feature takes precedence over the Directory Number Delayed Ringing feature (DNDR). If the MSB key is active on all ringing SCR/MCR sets, the non-ringing SCN/MCN sets ring immediately even if the DNDR feature is active.

If the MSB key is not active on all of the SCR/MCR sets, then the DNDR feature is applied to the SCN/MCN keys.

Executive Distinctive Ringing/Distinctive Ringing by DN

With the distinctive ringing features, the MSBI feature can assign different audible notifications to specific DNs. The audible notification is defined even if the DN is non-ringing. If the MSBI feature turns a non-ringing key into a ringing key, the defined distinctive audible notification is heard.

Multiple Appearance Directory Number

The Make Set Busy Improvement (MBSI) feature affects Multiple Appearance DNs, since the MSB key can manipulate the ringing or non-ringing of multiple appearance DNs.

Ringing Change Key

If Single Call Ringing (SCR) or Multiple Call Ringing (MCR) is changed to non-ringing by Ringing Change Key (RCK) and all ringing sets have MSB active, the sets ring immediately. If MSB is not active on all ringing sets, the lamp flashes on the non-ringing SCR or MCR.

If the SCN or the MCN key is changed from non-ringing to ringing, SCN and MCN lines are rung immediately. If one set is defined as ringing then a lamp flashes at non-ringing sets.

Short buzz for Digital sets

If the MSB key is activated on a set, and there is an incoming call to another SCN/MCN DN key on the same set, a buzzing (or short-buzzing) is applied immediately.

Private Line Service

If the MSB key is active on all ringing appearances of a Private Line DN, the Private Line non-ringing appearances of the same DN rings.

Feature packaging

The MSBI feature requires:

- The Make Set Busy (MSB) package 17

If analog(500/2500 type) sets are used, these additional packages are required:

- Background Terminal (BGD) package 99
- Flexible Features Codes (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 11 – Activate the MSBI feature and define Primary DN set (boss) with non-ringing DN key.
2. LD 11 – Define another set (secretary) with ringing DN key and MSB key.

LD 11 – Activate the MSBI feature and define Primary DN set (boss) with non-ringing DN key.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	 aaaa	Set type. Where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...	...	
CLS	MSIA	Allow Make Set Busy Improvement feature. (MSID) = Deny Make Set Busy Improvement feature.
...	...	
KEY	 xx SCN yyyy xx MCN yyyy xx PVN yyyy	Set function key assignments. xx SCN yyyy = Key number, Single Call Non-Ringing, DN. xx MCN yyyy = Key number, Multiple Call Non-Ringing, DN. xx PVN yyyy = Key number, Private Line Non-Ringing, DN.
...	...	

LD 11 – Define another set (secretary) with ringing DN key and MSB key.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	aaaa	Set type. Where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...	...	
KEY	 xx SCR yyyy xx MCR yyyy xx PVR yyyy xx MSB	Set function key assignments. xx SCR yyyy = Key number, Single Call Ringing, DN. xx MCR yyyy = Key number, Multiple Call Ringing, DN. xx PVR yyyy = Key number, Private Line Ringing, DN. xx MSB = Key number, Make Set Busy.
...	...	

Feature operation

No specific operating procedures are required to use this feature.

Malicious Call Trace

Contents

This section contains information on the following topics:

Feature description 809
Operating parameters 812
Feature interactions 813
Feature packaging 816
Feature implementation 817
Feature operation 824

Feature description

Malicious Call Trace (MCT) allows users of selected telephones to activate a call trace that results in a printed report of the calling and called parties. The report is generated on all system TTYs designated as maintenance (MTC) terminals.

Malicious Call Trace (MCT) is activated either by Dial Access from single-line (analog (500/2500 type) telephones), SL-1 and Meridian digital telephones (Meridian 1 proprietary telephones), or by key access from SL-1 telephones, Meridian digital telephones, and Attendant Consoles.

If the initiator hears overflow tone, the call trace has failed for one of the following reasons:

- The station does not have Malicious Call Trace Allowed (MCTA) Class of Service (CLS)

- The station is not established on an active call, or
- The system could not allocate a print register to store the trace information.

An attendant can activate Malicious Call Trace (MCT) only from an Attendant Console by using the Trace (TRC) feature key. When the Trace (TRC) key is pressed, the system prints a trace report on the source party, the destination party, or both, depending on whether the source key, the destination key, or both keys are active.

The MCT record identifies the source or destination (or both) by printing S or D (or both) prior to the time and date stamp of the record.

Enhanced Malicious Call Trace (EMCT)

With EMCT, the above feature provides the following enhancements:

- Malicious Call Trace is supported on Central Office (CO), Direct Inward Dial (DID) trunks.
- The alarm has a flexible ring timer, allowing a user-selectable range of from 0-15 minutes instead of being fixed at 15 minutes.
- The malicious call can be recorded by using a recording trunk.
- The call trace record can be printed on any Serial Data Interface (SDI) port when MCT is defined as a user. It is also written to the history file.

  Note: If MCT is not defined, the record is still printed on the maintenance TTY(s) only.

- The format of the call trace record tells you whether the call type is internal or external. The record identifier is either MCI for internal or MCE for external.

The user may configure an alarm to ring for a flexible period of time (0-15 minutes) for both internal and external calls. If the alarm DN goes off hook, it stops prior to the flexible alarm timer expiring.

Enhanced Malicious Call Trace for Saudi Arabia

From a user's perspective, the Malicious Call Trace feature activation remains the same as it was prior to this enhancement. However, with this enhancement the feature is now available for different types of analog and digital (CO, DID, and DOD) trunks. In order to send the MCT request, a special digit string is transmitted to the CO for an analog or digital trunk interface.

Enhanced Malicious Call Trace for Australia

In Australia, MCT can be activated during the established state of the call when interfaced with AXE-10 Australia on 2.0 Mbit Primary Rate Interface (PRI) trunks. MCT can also be activated during the call clearing state of the call (within a maximum of 30 seconds from the caller going on-hook). When MCT is activated, a special FACILITY message with a Key Pad information element is transmitted to the CO.

Trace Number (TRC) Key Lamp Status

The TRC key lamp status indicates the progress and success of the Malicious Call Trace request signaling to the CO and availability of the recorder. The following are the lamp states:

Lamp Winking

Activation of the TRC key changes the lamp from dark to winking (fast flashing) if the trunk involved in the call requires the signaling to be done. The lamp remains winking, indicating a transient state, until the call trace request signaling to the CO has been completed.

In a Meridian Customer Defined Network (MCDN) tandem scenario, the set which originated the call trace remains winking until a Facility message is received from the node nearest to the Central Office. The user cannot invoke MCT again while the lamp is in the winking state.

Lamp Lit

If the call trace request to the CO is successful and the recorder is conferenced in the call, the lamp state is changed to lit.

In an MCDN tandem scenario, the lamp goes from winking to lit if a Facility message received from the node nearest to the CO indicates that the MCT request was successful. Activation of the TRC key during this state is ignored.

Lamp Flashing

The lamp flashing (slower frequency than winkling) indicates that the call trace request to the CO was transmitted successfully, but a recorder could not be conferenced in. Activation of the TRC key during this state regenerates the MCT record, activates the alarm, and again attempts to conference in the recorder. The call trace request signaling to the Central Office is not transmitted again.

Lamp Dark

This lamp state indicates an idle TRC key or failure of the call trace request to the CO.

In an MCDN tandem scenario, the lamp goes from winking to dark if a Facility message received from the node nearest the CO indicates that the MCT request was unsuccessful.

Activation of the TRC during this state initiates all call trace elements again including: transmission of trunk hook flash; conferencing a recorder (if one is not already hooked in); generating an MCT record; and activating an alarm.

Operating parameters

The MCT feature is implemented on a system basis.

Assignment of the Trace (TRC) key cannot be done through the Attendant Administration feature.

The Enhanced MCT feature is available with all telephone types except BRI.

The TRC key cannot be assigned as a soft key on Meridian digital telephones.

Any country using flexible firmware flash timing (60-1536 msec.) requires the Generic XFCOT cards NTCK16AE or NTCK16BE, or the Extended Flexible Universal Trunk (EXUT) card NT8D14BA. For any country not using either the Generic Extended Flexible Central Office Trunk (XFCOT) card or the EXUT card, the same functionality is provided by software control.

The Multi-purpose Serial Data Link (MSDL) (or Downloadable D-channel for Small Systems must be used to support MCT for AXE-10 Australia (2.0 Mbit PRI).

MCT can be activated against only one established call at a time, regardless of the number of TRC keys defined.

The system is responsible for seizing the trunk to which recorders are connected. When a recorder is involved in the call, the call is treated as a conference call. The party on the source side is allowed to disconnect from the call; doing so also disconnects the recorder and resets the TRC key lamp to dark.

There is no special provision for warning tones while there is a conference with the recording device. The trunk is seized on the basis of the SRCH prompt in LD 16.

The following hardware is required to activate this feature on Large Systems: XFCOT card NTCK16AE, NTCK16BE; EXUT card NT8D14BA; 1.5 Mbps DTI interface QPC472E; 2.0 Mbps DTI interface QPC536B; PRI2 interface NT8D72AA; Digitone Receiver NT8D16AB; Tone and Digit Switch (TDS) NTAK03AA; Recorded telephone trunk; Conference card NT8D17CA; and MSDL card NT6D80AA. Note these are the minimum vintages required.

The following hardware is required for Small Systems: XUT NT8D14A; TDS/Digitone Receiver (DTR) NTAK03AA; CPU/CONF NAK01AA; 2.0 Mbps Primary Rate Interface (PRI) NTAK79AA; D-channel Handler (DCH) loadware NTBK50, NTBK51; 1.5 Mbps Digital Trunk Interface (DTI) NTAK09AA; 2.0 Mbps DTI NTAK10AA; Recorded telephone trunk NT8D14; and Generic XFCOT card NTCK16AE or NTCK16BE; and EXUT card NT8D14BA.

Feature interactions

Malicious Call Trace

China – Attendant Monitor

If a party involved in a monitored call activates the TRC key, monitoring is immediately deactivated.

Calling Party Privacy

Incoming calls to stations having the Malicious Call Trace feature enabled will continue to include the Terminal Number (TN) of the calling party in the Malicious Call Trace record, even if the caller has requested Calling Party Privacy.

Conference Call

When a station or console that is on the conference loop activates the MCT feature, the trace record shows only the conference loop number and conference number as the ORIGTN, and the Terminal Number (TN) of the station or console that activated the feature as the TERTN. No information on the other parties in the conference is given.

History File

The MCT records are stored in the History File if it has been defined as a maintenance (MTC) user in LD 17.

Meridian 911

The Malicious Call Trace (MCT) feature is modified to be supported on ACD sets. ACD sets are allowed to have the Malicious Call trace Allowed (MCTA) Class of Service and a Trace (TRC) key defined. The feature is activated by pressing the MCT key or dialing a MCT access code.

Meridian Mail

The Malicious Call Trace (MCT) feature is modified to be supported on Automatic Call Distribution (ACD) sets. ACD sets are allowed to have the MCTA Class of Service and a TRC key defined. The feature is activated by pressing the MCT key or dialing a MCT access code.

Enhanced Malicious Call Trace

Autodial Tandem Transfer

Enhanced Malicious Call Trace implements the ability to send a call trace request to the CO and provides the possibility to record the call using a recorder. This feature also uses the Centrex/Trunk Switchhook Flash feature; the same enhancement applies to the Autodial Tandem Transfer feature.

Automatic Call Distribution (ACD) Emergency Key (EMR)

The Malicious Call Trace feature operates in a similar manner to the Automatic Call Distribution (ACD) Emergency Key (EMR) feature when conferencing a recording. In this enhancement, the ACD set can activate both the Malicious Call Trace and ACD EMR features.

Called Party Control Option

Prior to this feature, the Called Party Control (CDPC) option was not supported for conference calls. The CDPC option is now supported if the conference contains exactly one recording trunk, one MCT activating party and one other trunk. This is done to make the recorder transparent to the user. The CDPC option remains unsupported for all other conference calls.

Centrex Switchhook Flash

Interaction with the Centrex switchhook flash results because the flash range is changed for this feature. Communication to the CO (trunk hook flash) is performed by using the Centrex switchhook flash feature base code. The enhanced range is available for the Centrex switchhook flash.

Collect Call Blocking

If a station activates Malicious Call Trace (MCT) while the Collect Call Blocking answer signal is being sent, MCT activation is ignored. This also applies to the case when MCT is activated from a remote node.

Conference Call

If MCT is activated during a conference, the trace record shows the conference number and the conference loop number. Trace records are printed for each party involved in the conference. The originator of the call's trace record is printed first.

History File

If the SDI port is defined as an MCT user in LD 17 or the SDI port as a maintenance (MTC) user in LD 17, the MCT records can be stored in the History File. If MCT and MTC users are both defined on the TTY in LD 17, MCT records can also be stored in the History File.

Malicious Call Trace DN/TN Print

If the option MCDC (in LD 15) is set, a second line is added in the MCT reports to show the DN of both parties of the call. If Calling Line Identification (CLID) is available, it is printed in the second line.

Malicious Call Trace Idle Signal

The existing operation of the Malicious Call Trace Idle Signal feature is unchanged.

Meridian 911

The Trunk Hook Flash functionality is used by Meridian 911, Enhanced Malicious Call Trace, and Autodial Tandem Transfer.

Feature packaging

Malicious Call Trace (MCT) and Enhanced Malicious Call Trace (EMCT) require Malicious Call Trace (MCT) package 107.

For ISDN environments, ISDN packages are required based on the node and network interface applicable to the specific country.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable MCT on an Analog (500/2500 type) telephone.

2 LD 11 – Enable MCT on a Proprietary Telephone.

3 LD 17 – Allow printing of the MCT record on a dedicated MCT TTY port.

4 LD 16 – Set up the recorder route.

5 LD 14 – Set up the recorder trunk.

6 LD 15 – Set up the recorder and alarm options.

7 LD 16 – Set up the alarm for external calls.

8 LD 57 – Define the MCT FFC.

9 LD 16 – Configure the call trace string.

10 LD 14 – Enable Firmware timing for trunk hook flash (if available).

11 LD 73 – Define the DTI2 flash time range.

12 LD 16 – Set up MCTM timer and tandem delay (2 Mbps PRI for AXE-10 Australia only).

Note: In order to activate Malicious Call Trace from an analog (500/2500 type) telephone, the user has to dial SPRE + two-digit access code (83) or the MCT Flexible Feature Code FFC.

LD 10 – Enable MCT on an Analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Analog (500/2500 type) telephone data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	MCTA, MCTD	Malicious Call Trace is allowed if Class of Service is MCTA.

Note: In order to activate Malicious Call Trace from a Meridian 1 proprietary telephone, it should have CLS MCTA, and the TRC key should be defined. However, the same function can be achieved using a transfer or conference key and the SPRE + 83 or the MCT FFC.

LD 11 – Enable MCT on a Proprietary Telephone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Meridian 1 proprietary telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(MCTD) MCTA	Malicious Call Trace is allowed if Class of Service is MCTA.
...		
KEY	xx TRC	Key number; Malicious Call Trace. Allowed when CLS = MTA. Key lamp not required. MCT is applied on a TN basis. This key can be configured on ACD telephones.

LD 17 – Allow printing of the MCT record on a dedicated MCT TTY port.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	Configuration Record. Gate opener.
- ADAN	xxx TTY yy	xxx = NEW or CHG. yy = port number 0-63 or 0-15.
- USR	MCT	Dedicated TTY port for MCT record.

LD 16 – Set up the recorder route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route data block.
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RCD	Recorder trunk data block.
ACOD	xxxx	Recorder route access code.

LD 14 – Set up the recorder trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RCD	Recorder trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

CUST	xx	Customer number, as defined in LD 15
RTMB	0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems

LD 15 – Set up the recorder and alarm options.

Prompt	Response	Description
REQ:	CHG PRT END	Change, print, or end.
TYPE:	FTR	Features and options
CUST	0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- ALDN	xxxxxxx	DN for the alarm (the DN must be on the local system).
- ALRM	(NO) YES	The ALRM prompt appears only if ALDN is defined. ALRM has to be set to YES if the alarm is to be rung for any call (external or internal) when MCT is activated.
- TIME	0-(15)	Time is prompted only if ALRM is set to YES. Time for the alarm is set in one-minute increments from 1 to 15.
- INT	(NO) YES	INT is prompted only if ALRM is set to YES. In addition, INT must be YES if the alarm is to be rung when MCT is activated against internal calls.
- RECD	(NO) YES	If the user wants the recorder, set RECD to YES. This prompt does not appear when a new customer is being defined.
- - MCRT	xxxx	The user has to use the recorder route number defined in LD 16. It will only be prompted if the RECD is set to YES.

LD 16 – Set up the alarm for external calls.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route data block.
TKTP	DID COT	Direct Inward Dial or Central Office trunks.
ALRM	(NO) YES	Malicious Call Trace is allowed for external calls when the response is YES.

Note: In order to activate Malicious Call Trace from an analog (500/2500 type) telephone without using the SPRE and 83, the MCT FFC has to be defined.

LD 57 – Define the MCT FFC.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15
CODE	MTRC	Malicious Call Trace.
MTRC	xxxx	Flexible Feature Code for Malicious Call Trace.

Note: For analog and 1.5 Mbps digital trunks, the flash range to be sent to the Central Office is configured using the FLH timer. In order to send the string to the Central Office, MCCD has to be defined

LD 16 – Configure the call trace string.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.

TYPE	RDB	Route data block.
RCLS	(EXT) INT	Class marked route as (external) or internal.
...		
CNTL	YES	Changes control or timers.
- TIMR	FLH <space> 60-(510)-32640	Hook Flash timer (in msec.) The range for Centrex Switchhook flash timer is 256-(512)-1536. For CAS, it is recommended that the timer be set at 768 or greater. This timer must be at least 256 ms shorter than the remote OGF timer and 256 ms shorter than the ICF timer. • 60-89 ms = Digit 1 is sent • 90 ms = Hard coded for XFCOT hook flash • 91-255 ms = Digit 1 is sent • 256-1536 ms = Existing software controlled hook switch flash Range for Centrex Switchhook flash timer is 60-(510)-1536 msec (the value is rounded to the nearest 10 msec). Software controlled Centrex/Trunk Switch Flash timer range of 60- 127 msec is done by sending digit 1. The range of 128-1536 msec is already controlled by Centrex Switchhook Flash feature. Firmware flash user can enter any value from 60 to 1536. FWTM must be YES in LD 14 for the trunk associated with this route, if firmware timing is to be used.
...		
MCTS	(NO) YES	Enter YES to get the new prompts
MCCD	0-8 digits	The call trace request string can be 0-8 digits in length. Valid digits are 0-9, *, and #.
MCDT	(0)-4	Digit string delay is in seconds, in increments of one second.

Note: The FWTM prompt is provided for EXUT and XCOT cards. This prompt should be set to YES if firmware timing is to be done for the flash and the card supports this functionality. If the prompt is set to YES for one unit, it is also set to YES for all other units.

LD 14 – Enable Firmware timing for trunk hook flash (if available).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DID COT	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	EXUT XCOT	Card type
FWTM	(NO) YES	Firmware timing for the trunk hook flash is available. This prompt is set to YES if firmware timing for trunk hook flash is supported by the card.
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems

LD 73 – Define the DTI2 flash time range.

Prompt	Response	Description
REQ	NEW CHG PRT	New, change, or print.
TYPE	DTI2	
FEAT	abcd	Digital signaling category.
SICA	2-16	SICA table number.
...		

FALT (R)	abcd N	Received bits. If FALT (receive) signal is not required.
P RRC(S)	abcd	Register recall signal activated by MCT.
TIME	10-(100)-630	Time of RRC(S) signal in milliseconds. This is the flash duration used for 2.0 Mbit DTI trunks. It is programmable in one-millisecond increments from 10 to 630.

LD 16 – Set up MCTM timer and tandem delay (2 Mbps PRI for AXE-10 Australia only).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	CDB	Customer data block.
CUST	xx	Customer number, as defined in LD 15
MCTS	YES NO	
MCTM	(0) - 30	Malicious Call Trace timer (in seconds).
MTND	(NO) YES	Malicious Call Trace disconnect delay for tandem calls for AXE-10 Australia.

Feature operation

To trace a malicious call from an analog (500/2500 type) telephone:

1. Flash the switchhook. A special dial tone signifies that the call is on hold.
2. Enter SPRE+83. You are reconnected to the call.

To trace a malicious call from a Meridian 1 proprietary telephone using Special Prefix (SPRE) code:

1. Press **Transfer** or **Conference**. A special dial tone signifies that the call is on hold.
2. Enter SPRE+83. You are reconnected to the call.

To trace a malicious call from a Meridian 1 proprietary telephone using the Trace (TRC) key:

1 Press **Call Trace**. You remain connected to the call.

Malicious Call Trace DN/TN Print

Contents

This section contains information on the following topics:

Feature description 827
Operating parameters 827
Feature interactions 827
Feature packaging 828
Feature implementation 828
Feature operation 828

Feature description

This feature enhancement adds a second line to the Malicious Call Trace (MCT) record, printed on the maintenance TTY. This second line provides information about the DNs of the calling and called parties. For trunk calls, the Calling Line Identification (CLID) number (if available) is printed. This enhancement does not change the functionality of the Malicious Call Trace feature.

Operating parameters

See the “Malicious Call Trace” feature in this document.

Feature interactions

See the “Malicious Call Trace” feature in this document.

Feature packaging

Malicious Call Trace (MCT) package 107.

Feature implementation

LD 15 – Enable Printing of Malicious call DN/CDIP information.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
...		
- MCDC	YES	Allow the printing of Malicious Call DN/CLID information for the originating and terminating parties.

Feature operation

The modified MCT record output format is as follows:

First Line

Field No.Field TypeContents

1 Record TypeMCT++

2 Customer No.CUSTxx++

3 Originator<*>TNlscu++/<*>TNlc++/<*>CFlc++

4 Terminator<*>TNlscu++/<*>TNlc++/<*>CFlc++

5 Source/Dest.+/S/D++

6 Time stamp hh:mm:ss++MM/DD/YYYY

7 CNIDCNI#xxxxxxxxxxxxxxx

Second Line

Field No.Field TypeContents

1 Originator<*>DNxxxxxxx+++++++++++++++

2 Terminator<*>DNxxxxxxx+++++++++++++++

or it could be of the following combinations of a DN and CLID number:

1 Originator<*>CLID#xxxxxxxxxxxxxxx++

2 Terminator<*>DNxxxxxxx+++++++++++++++

or

1 Originator<*>DNxxxxxxx+++++++++++++++

2 Terminator<*>CLID#xxxxxxxxxxxxxxx++

Malicious Call Trace Idle

Contents

This section contains information on the following topics:

Feature description 831
Operating parameters 831
Feature interactions 832
Feature packaging 832
Feature implementation 833
Feature operation 835

Feature description

The Malicious Call Trace (MCT) Idle signal instructs the Public Exchange/ Central Office to give the called party control of the call connection. If the called party does not go on-hook at the end of a conversation, the connection will be held through the Public Switched Telephone Network (PSTN) indefinitely by means of a Multifrequency Compelled (MFC) Idle Call Trace (IDCT) signal generated by the system. This feature allows the automatic call-tracing equipment in the PSTN to print out the appropriate details of the calling party.

Operating parameters

Direct Inward Dialing (DID) calls which terminate on idle trunks result in the IDLE signal being returned to the Central Office.

DID calls which terminate on an Attendant Console result in either a Multifrequency Compelled IDLE or IDCT signal being returned, depending on the customer option. This applies to both direct and intercept calls.

When an Attendant Console is in Night Service, the signal being returned is determined by the customer option and not by the classification of the night DN, unless a DID call comes into a night DN.

When a DID call is diverted prior to termination, either by Call Forward, Hunting, or Call Forward Busy, the signal being returned is determined by the called party extension classification.

If a DID call terminates at a Multiple Appearance DN in which at least one station has malicious call trace allowed Class of Service, then a Multifrequency Compelled IDCT signal is returned to the Central Office. If all stations sharing the DN have Malicious Call Trace denied Class of Service, a Multifrequency Compelled IDLE signal is returned.

Feature interactions

Malicious Call Trace - Enhanced

The existing operation of the Malicious Call Trace Idle Signal feature is unchanged.

Recorded Announcement for Calls Diverted to External Trunks

DID calls to a busy Recorded Announcement (RAN) trunk group are queued and receive ring-back tone. A Multifrequency Compelled IDLE signal is returned.

Trunk Supervision

Once a Multifrequency Compelled IDCT signal is returned, the disconnect trunk supervision is limited to the called party.

Feature packaging

Malicious Call Trace (MCT) package 107.

Dependency:

- Multifrequency Compelled Signaling (MFC) package 128.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable MCT on an Analog (500/2500 type) telephone.

2 LD 11 – Enable MCT on a proprietary telephone.

3 LD 15 – Enable the MCT signal.

4 LD 94 – Create or modify the MFC tables.

5 LD 16 – Create or modify data for each DID trunk route data block to allow or deny MFC Signaling option.

6 LD 14 – Create or modify data for each DID trunk data block to allow or deny MFC Signaling option.

LD 10 – Enable MCT on an Analog (500/2500 type) telephone.

Prompt	Response	Description
CLS		Class of Service.
	(MCTD) MCTA	Malicious Call Trace (denied) allowed. When MCTA is assigned, the station must also have XFA defined.
	(XFD) XFA	Call Transfer (denied) allowed.

LD 11 – Enable MCT on a proprietary telephone.

Prompt	Response	Description
CLS		Class of Service.
	(MCTD) MCTA	Malicious Call Trace (denied) allowed. When MCTD is assigned, the MCT key is removed.
KEY	xx TRC	MCT Key number.

LD 15 – Enable the MCT signal.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
...		
- OPT	MCTA	Malicious Call Trace signal is allowed for attendants at this customer location.

LD 94 – Create or modify the MFC tables.

Prompt	Response	Description
TYPE	MFT	Multifrequency table.
ICOG	ICT OGT	Incoming Table, Outgoing Table.
TBNO	1 - 127	MFC Table number.
XMIT	IDCT n	Idle Call Trace Signal number.

LD 16 – Create or modify data for each DID trunk route data block to allow or deny MFC Signaling option.

Prompt	Response	Description
MFCI	(0) - 127	MFC Incoming table number.
AUTO	NO	Auto terminate.
MFCO	(0) - 127	MFC Outgoing table number.
AUTO	YES	Auto terminate.
CDCT	(NO) YES	Called Party Control (is not) is to be allowed on Malicious Call Trace Idle Calls.
CDPC	(NO) YES	Called Party Control (is not) is activated when the IDCT signal is sent for non-toll calls.

LD 14 – Create or modify data for each DID trunk data block to allow or deny MFC Signaling option.

Prompt	Response	Description
CLS		Class of Service.
	(DIP) DTN MFC	Dial Pulse. Dual Tone Multifrequency. R2 MFC Signal.
MFL	(0) - 7	MFC digit level required for signals to PSTN.

Feature operation

No specific operating procedures are required to use this feature.

Malicious Call Trace on Direct Inward Dialing

Contents

This section contains information on the following topics:

Feature description 837
Operating parameters 837
Feature interactions 838
Feature packaging 838
Feature implementation 838
Feature operation 840

Feature description

This feature provides an enhancement to the Malicious Call Trace (MCT) feature. If the MCT feature is activated by pressing the trace (TRC) key (on a Meridian 1 proprietary telephone or Attendant Console), or by dialing the SPRE and 83, a digit 1 is outpulsed to the trunk. This is an indication to the Public Switched Telephone Network (PSTN) to activate its own MCT feature.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the Malicious Call Trace on DID feature described above.

The Malicious Call Trace on DID feature is not available on 1.5 Mbit digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 10 – Enable MCT on an analog (500/2500 type) telephone data block.
2. LD 11 – Enable MCT on a Meridian 1 proprietary telephone data block.
3. LD 15 – Enable MCT signal.
4. LD 94 – Create or modify the MFC tables.

LD 10 – Enable MCT on an analog (500/2500 type) telephone data block.

Prompt	Response	Description
...		
CLS		Class of Service.
	MCTA	Malicious Call Trace allowed. When MCTA is assigned, the station must also have XFA defined.
	(XFD) XFA	Call Transfer (denied) allowed.

LD 11 – Enable MCT on a Meridian 1 proprietary telephone data block.

Prompt	Response	Description
...		
CLS		Class of Service.
	MCTA	Malicious Call Trace allowed. When MCTD is assigned, the MCT key is removed.
KEY	xx TRC	MCT Key number.

LD 15 – Enable MCT signal.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	FTR	Features and options
...		
OPT	MCTA	Malicious Call Trace signal is allowed for attendants at this customer location.

LD 94 – Create or modify the MFC tables.

Prompt	Response	Description
...		
TYPE	MFT	Multifrequency table.
ICOG	ICT OGT	Incoming Table, Outgoing Table.
TBNO	1-127	MFC Table number.
XMIT	IDCT n	Idle Call Trace Signal number.

Feature operation

No specific operating procedures are required to use this feature.

Manual Line Service

Contents

This section contains information on the following topics:

Feature description 841
Operating parameters 841
Feature interactions 842
Feature packaging 842
Feature implementation 843
Feature operation 843

Feature description

Manual Line Service allows all calls made from an analog (500/2500 type) telephones defined as manual telephones to be handled automatically by an attendant. When the caller goes off-hook, the attendant is contacted immediately. Calls can be placed to telephones with Manual Line Service.

Operating parameters

Manual Line Service applies only to analog (500/2500 type) telephones.

Feature interactions

Attendant Alternative Answering

When Attendant Alternative Answering (AAA) is defined, Manual Line Service follows the AAA parameters.

Attendant Overflow Position

When Attendant Overflow Position (AOP) is defined, Manual Line Service follows the AOP directions.

Automatic Wake Up

Automatic Wake Up (AWU) does not support these features; an AWU call cannot be programmed against a manual line or private line DN.

Night Service

When the system is in Night Service (NSVC) mode, all telephones with a manual Class of Service are routed to the telephone designated as the night number for the customer group.

Phantom Terminal Numbers

Manual Line Service cannot be enabled on a phantom terminal number.

Station-to-Station Calling

If a single line telephone has been assigned a Manual Line Class of Service, the telephone automatically rings the attendant when it goes off-hook.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 10 – Define Class of Service for Manual Line telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DN	xxx...x	Directory Number assigned to the telephone.
CLS	MNL	Arrange telephone for Manual Line Service.

Feature operation

No specific operating procedures are required to use this feature.

Manual Service Recall to Attendant

Contents

This section contains information on the following topics:

Feature description 845
Operating parameters 845
Feature interactions 846
Feature packaging 846
Feature implementation 846
Feature operation 846

Feature description

This feature allows an incoming Direct Inward Dialing (DID) trunk with far-end control, that has been disconnected at the system end, to perform an attendant recall upon receiving a switchhook flash.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Manual Service Recall to Attendant feature described above.

The Manual Service Recall to Attendant feature is not available on 1.5 Mbit digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify data for each DID trunk route data block to have or deny MFC Signaling:

Prompt	Response	Description
...		
RCAL	(NO) ATT	Enter ATT to allow Manual Service Recall to the attendant.

Feature operation

To perform an attendant recall upon flash the switchhook. The switchhook flash is considered valid if it lasts at least 30 milliseconds.

When the switchhook flash signal is recognized by the system as being valid, the call is immediately presented to the attendant or to the Night Service number if the attendant is in Night Service.

Manual Signaling (Buzz)

Contents

This section contains information on the following topics:

Feature description 847
Operating parameters 848
Feature interactions 848
Feature packaging 848
Feature implementation 849
Feature operation 849

Feature description

Manual Signaling (Buzz) permits a Meridian 1 proprietary telephone user to sound a buzz tone at a specific telephone. The Meridian M3000 Touchphone provides the buzzing capability by means of an Active State screen softkey.

To activate this feature, a separate buzz key must be equipped. An associated lamp or indicator is not required.

The buzz tone continues as long as the key remains depressed. Manual Signaling (Buzz) has no impact on an existing call or on other active features. If the other telephone is busy on a call, it will still buzz, even if it is a Handsfree call.

Operating parameters

Manual Signaling (Buzz) does not apply to analog (500/2500 type) telephones.

Only Single Appearance Directory Numbers can be buzzed.

Feature interactions

Call Party Name Display

If the Signal key is pressed to buzz another telephone, no digit or name display appears on the telephone.

Network and Executive Distinctive Ringing

Network Distinctive Ringing and Executive Distinctive Ringing do not affect the buzzing of a set.

Voice Call

The same DN can be used for both Voice Call and Manual Signaling (Buzz) as long as it remains a Single Appearance DN.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Add Manual Signaling (Buzz) key for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx SIG yyy...y	Add a Manual Signaling (Buzz) key, where: xx = key number, and yyy...y = DN to be buzzed (must be a Single Appearance Directory Number).

Feature operation

To buzz a specific telephone:

- Press **Buzz**. The other telephone emits a buzz sound from the speaker for as long as you hold down the Buzz key.

Manual Trunk Service

Contents

This section contains information on the following topics:

Feature description 851
Operating parameters 851
Feature interactions 852
Feature packaging 852
Feature implementation 852
Feature operation 854

Feature description

Manual outgoing trunk service permits you to complete an outgoing call, after ringing the trunk, by dialing a predefined trunk access code. Manual incoming trunks, when seized at the far end, are automatically terminated on a specified Directory Number (DN) or, if no DN is specified, at the attendant.

Manual Trunk Service is defined by the trunk Class of Service, and can be applied to outgoing, incoming, and outgoing/incoming trunks. This feature is available to the Central Office (CO), FX, WATS, and TIE trunks with an immediate start arrangement.

Operating parameters

Manual incoming service can be applied to TIE trunks only.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Add or change an incoming manual trunk route.

2 LD 14 – Add or change an incoming manual trunk.

3 LD 16 – Add or change an outgoing manual trunk route.

4 LD 14 – Add or change an outgoing manual trunk.

LD 16 – Add or change an incoming manual trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE	Incoming manual trunks (must be TIE trunks).
ICOG	ICT	Incoming route.
ACOD	xxxx . . x	Trunk route access code.

LD 14 – Add or change an incoming manual trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE	TIE trunks are required for manual incoming trunks.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
MNDN	xxx...x	Directory Number for automatically terminate.
SIGL	aaa	Trunk signaling, where: aaa = DX2, DX4, EAM, EM4, GRD, LDR, LOP, or OAD.
STRI	IMM	Incoming start arrangement.
SUPN	(NO) YES	Answer and disconnect supervision (not required) or required.
CLS	(MID) MIA	Manual incoming service (denied) allowed.

LD 16 – Add or change an outgoing manual trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15

ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	aaa	Outgoing trunk type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, ISL, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
ICOG	OGT	Outgoing route.
ACOD	xx . . x	Trunk route access code.
MANO	YES	Enable manual outgoing trunk route.

LD 14 – Add or change an outgoing manual trunk.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaa	Outgoing trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
MNDN	xx...x	Directory Number for automatically terminate.
SIGL	aaa	Trunk signaling, where: aaa = DX2, DX4, EAM, EM4, GRD, LDR, LOP, or OAD.

Feature operation

No specific operating procedures are required to use this feature.

Meridian 1 Attendant Console Enhancements

Contents

This section contains information on the following topics:

Feature description 855
Operating parameters 870
Feature interactions 875
Feature packaging 880
Feature implementation 880
Feature operation 883

Feature description

The Meridian 1 Attendant Console Enhancements (MACE) feature expands existing Meridian 1 Attendant Console functionality. This feature provides the following enhancements:

- Attendant Console Autoline
- Individual Attendant Console Directory Number (IADN)
- Attendant Emergency Codes

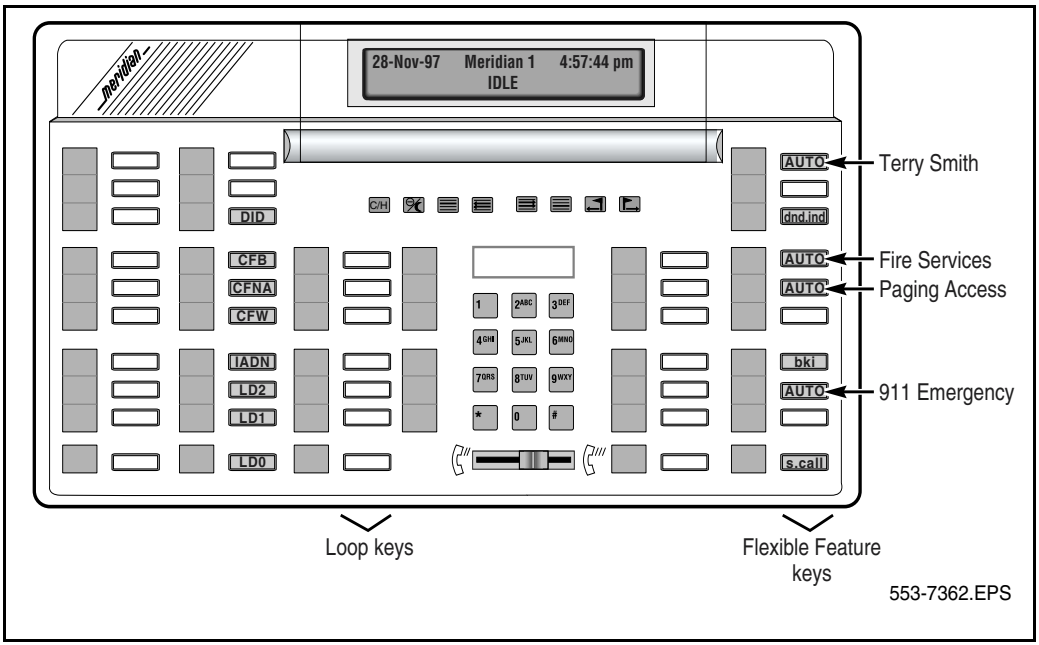
Attendant Console Autoline

The Attendant Console Autoline functionality provides secure autodial services for all types of Attendant Consoles. These services are programmed on Flexible Feature keys on an attendant basis. When the Autoline key is activated, the system automatically dials a pre-programmed Directory Number (DN). The DN that is stored for the Autoline key can be from 1-31 digits in length and can be either internal or external to the system.

The Autoline key's functionality is almost identical to that of the Autodial key. However, with Autoline functionality, the DN cannot be programmed from the console. Also, the display key function is simplified. With the Autoline functionality, to display a DN programmed for the Autoline key, the attendant presses the Autoline key when the console is idle or in Position Busy. On an analog console, to display a DN that is longer than eight digits, the attendant presses the Display Source key after pressing the Autoline key.

Figure 19 illustrates an Attendant Console with four Autoline keys configured on Key Strip 5. This key strip holds the Flexible Feature keys. On Key 2, Autoline is configured to dial 911 for Emergency Calls; on Key 5, Autoline is configured for Paging Access; on Key 6, Autoline is configured to dial Fire Services; and on Key 9, Autoline is configured to dial Terry Smith's DN.

Figure 19
Attendant Console with four Autoline keys configured



In order for an Autoline call to be placed, the attendant presses a Loop key and then presses the Autoline key. When the Autoline key is pressed, the pre-programmed number is automatically dialed.

Figure 20 shows an Attendant Console display when an Autoline call is placed. In this example, the attendant places an Autoline call to Terry Smith at DN 2029. The attendant presses the Loop key and then the Autoline key that is configured to dial Terry Smith's DN. In this case, once the Autoline key is pressed, the attendant display is as shown in Figure 20.

Figure 20
Attendant display when the Autoline functionality is in progress

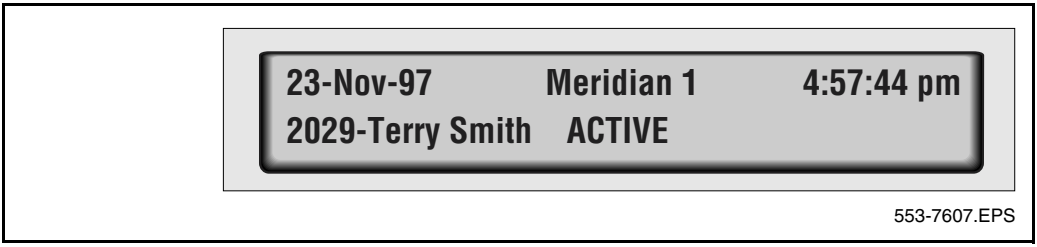
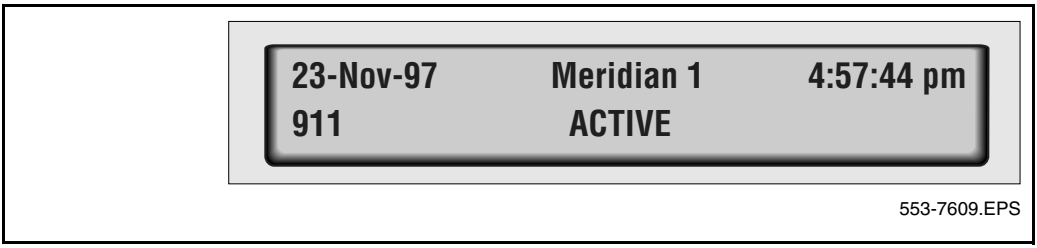


Figure 21 shows an Attendant Console display when an Autoline call is placed to an Autoline DN that is external to the system. In this example, the Autoline key is programmed for 911 Emergency. When the attendant presses the Loop key and then the Autoline key, the display shows the external DN that is programmed for the Autoline key. In this case, the external DN is 911.

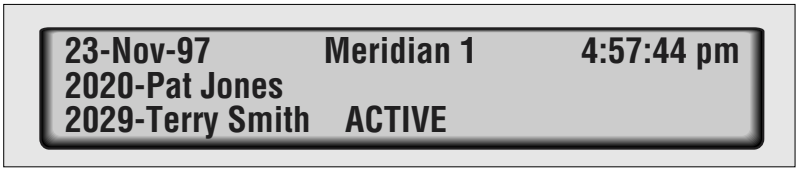
Figure 21
Attendant display when the attendant places an Autoline call to an external Autoline DN



If an attendant is already active on a call and wishes to extend that call to the Autoline DN, the Autoline key is pressed to extend the call.

Figure 22 shows an example of an Attendant Console display when the attendant is already involved in an established call. In this example, Pat Jones at DN 2020 dials the attendant, and a call is established. The attendant wishes to extend the current call to Terry Smith at DN 2029 and does so by pressing the Autoline key that is configured with Terry Smith's DN. Once the Autoline key is pressed, the attendant display is as shown in Figure 22. When the attendant presses the Release key, the display is cleared.

Figure 22
Attendant display when the attendant extends a call to the Autoline DN



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Individual Attendant Console Directory Number (IADN)

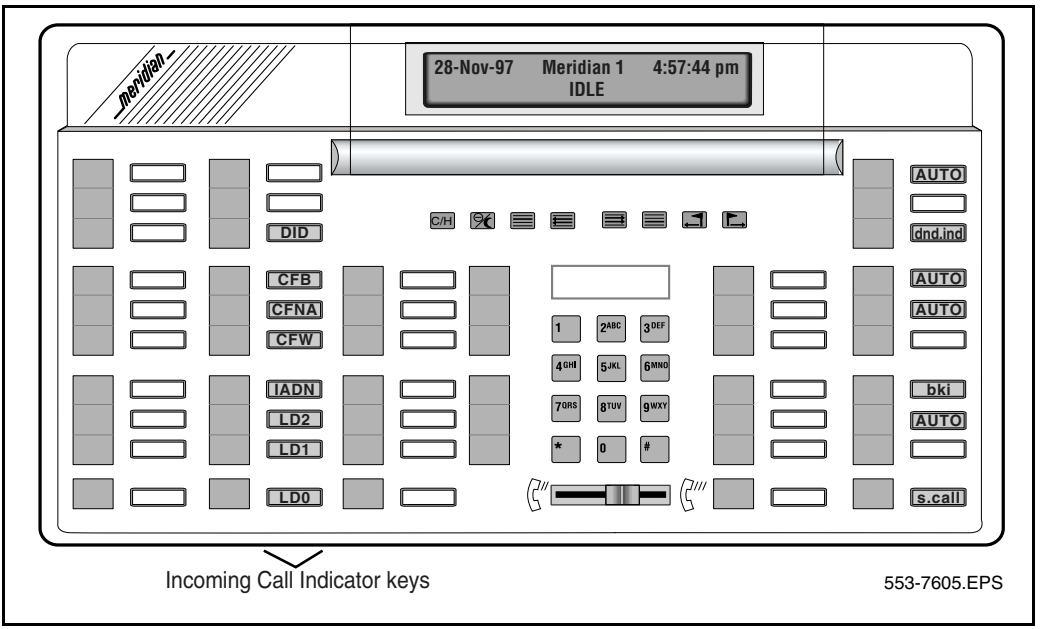
The Individual Attendant Console Directory Number (IADN) functionality allows digital attendant consoles (M2250) to be directly contacted from an internal or external set. Individuals who are paged by an Attendant Console can now re-call that specific console directly, using a new DN type - Individual Attendant Console Directory Number (IADN).

The IADN can have a maximum of four digits or seven digits if Directory Number Expansion (DNXP) package 150 is equipped. The IADN is defined at an attendant level. For an external set to reach the IADN console, the IADN must be defined as a Direct Inward Dialing (DID) number.

A new Incoming Call Indicator (ICI) key is also introduced with the IADN functionality. The IADN ICI key is defined at a customer level. It allows the attendant to answer an IADN call "out of turn" from the attendant queue. If there is at least one IADN call waiting in the attendant queue, the IADN ICI key lamp flashes.

Figure 23 illustrates a Meridian 1 Attendant Console with an IADN ICI key configured on Key Strip 2. This key strip holds the ICI keys.

Figure 23
Attendant Console with an IADN ICI key configured



When an IADN call is made to an Attendant Console that is already active, the call is placed in the attendant queue. An audible tone, Priority Buzzing, may be provided to the active attendant as an indication that an IADN call is waiting to be answered.

> ***Note:*** The system does not place IADN calls ahead of other calls in the attendant queue. It is the attendant who gives priority to IADN calls by answering them on the IADN ICI key.

For Priority Buzzing to be provided to the active attendant, the Individual Attendant DN Buzzing (IDBZ) prompt must be set to YES in the Customer Data Block. Also, the IADN ICI key must be configured by defining the Incoming Call Indicator (ICI) prompt in the Customer Data Block.

The default cadence for Priority Buzzing is two seconds on and ten seconds off. However, the cadence can be modified with the Priority Buzzing Cadence (PBUZ) prompt in the Customer Data Block. The PBUZ prompt is a prompt introduced with this feature.

The flexible cadence value range is from 2-16 seconds in multiples of two seconds for the on and the off buzzing phases. If the value entered for either of these two phases is an odd number in the valid range, it is rounded down. For example, if the value entered for the on or off buzzing phase is five, it is rounded down to four.

Idle Attendant Console

An Attendant Console is idle when it is available to receive incoming calls. When an internal or external party dials the idle attendant's IADN, the call is presented to the attendant on an idle Loop key. The IADN ICI key lamp, if configured, flashes when the call is presented, and the Attendant Console receives a continuous buzz. Hence, Priority Buzzing is not applicable in this case. When the attendant answers the call, the IADN ICI, Source (SRC), and Loop key lamps are all lit on the console.

Active Attendant Console

When an Attendant Console is in an active state, the Release (RLS) key lamp is not lit. When an internal or external set places a call to the active IADN attendant, the call waits in the attendant queue to be answered. The treatment given to such a call depends upon whether or not the IADN ICI key is configured as well as how the IDBZ prompt is defined in the Customer Data Block.

When an IADN ICI key is configured and the IDBZ prompt is set to NO in the Customer Data Block, Priority Buzzing is **not** provided when an IADN call is waiting to be answered in the attendant queue. Consider the following example:

1 An IADN attendant is involved in an active call.

2 An IADN call is placed to the active attendant and waits to be answered in the attendant queue. No Priority Buzzing is provided to the Attendant Console.

3 The attendant releases the active call.

4 The next call in the queue is presented to the attendant. All ICI keys on the Attendant Console, including the IADN key, are updated. The IADN ICI key lamp flashes if there is at least one IADN call waiting in the attendant queue.

5 The attendant chooses to answer the IADN call, from the queue, by pressing the IADN ICI key.

When an IADN ICI key is configured and the IDBZ prompt is set to YES in the Customer Data Block, Priority Buzzing is provided when an IADN call is waiting to be answered in the attendant queue. Consider the following example:

1 An attendant is involved in an active call.

2 An IADN call is placed to the active attendant and waits to be answered in the attendant queue.

3 Priority Buzzing is provided to the Attendant Console. During this time, if another IADN call for the same attendant, is placed in the attendant queue, the Priority Buzzing is not affected.

4 The attendant releases the active call.

5 The next call in the queue is presented to the attendant.

6 The Priority Buzzing stops, and the attendant receives a continuous buzz for the newly presented call. All ICI keys on the Attendant Console, including the IADN key, are updated. The IADN ICI key lamp flashes if there is at least one IADN call waiting in the attendant queue.

7 The attendant chooses to answer the IADN call, from the queue, by pressing the IADN ICI key. If there is another IADN call waiting for the attendant in the queue, Priority Buzzing is applied to the attendant again. If there is not another IADN call waiting, then the Priority Buzzing stops. If the attendant selects another call over the IADN call (using another ICI key or taking a non-IADN call if presented on the Loop key), Priority Buzzing begins again.

When an IADN ICI key is **not** configured and whether or not the IDBZ prompt is set to YES, Priority Buzzing **does not** function. The IADN ICI key must be configured for the Priority Buzzing functionality to be applicable. Consider the following example:

1 An attendant is involved in an active call.

2 An IADN call is placed to the active attendant and waits in the attendant queue. No Priority Buzzing is provided to the Attendant Console.

3 The attendant releases the active call.

4 The next call in the queue is presented to the attendant.

5 The IADN call is only presented to the attendant when its “turn” comes about in the attendant queue. The IADN call is presented on a Loop key in this case.

Attendant Console in Position Busy

An Attendant Console is not able to receive incoming calls when it is in a Position Busy state. In this case, IADN calls unable to reach the busy attendant are treated as normal attendant calls and are instead sent to an available Attendant Console in the system. Priority Buzzing is not provided to the available console, and the IADN ICI key does not flash, as the IADN call was not originally intended for this particular console. The IADN call is presented to the attendant on a Loop key when its “turn” comes about in the attendant queue. No ICI keys are lit for these calls.

When an IADN console leaves the Position Busy state, it receives Priority Buzzing for all of the IADN calls waiting in the attendant queue.

Customer/Tenant in Night Mode

A customer or tenant is in Night Mode when all of its Attendant Consoles are in Position Busy. When an IADN call is placed to an Attendant Console in this situation, the call receives the standard night treatment defined for the Customer. If Network Attendant Service (NAS) is equipped and also has NLDN, Priority Buzzing is provided (if configured).

An Attendant Console returns to an idle state from Position Busy with an IADN call waiting in the attendant queue. Priority Buzzing is only provided to this Attendant Console if there is more than one call waiting in the attendant queue and if the IADN call is not the first call in queue. Otherwise, normal attendant treatment occurs.

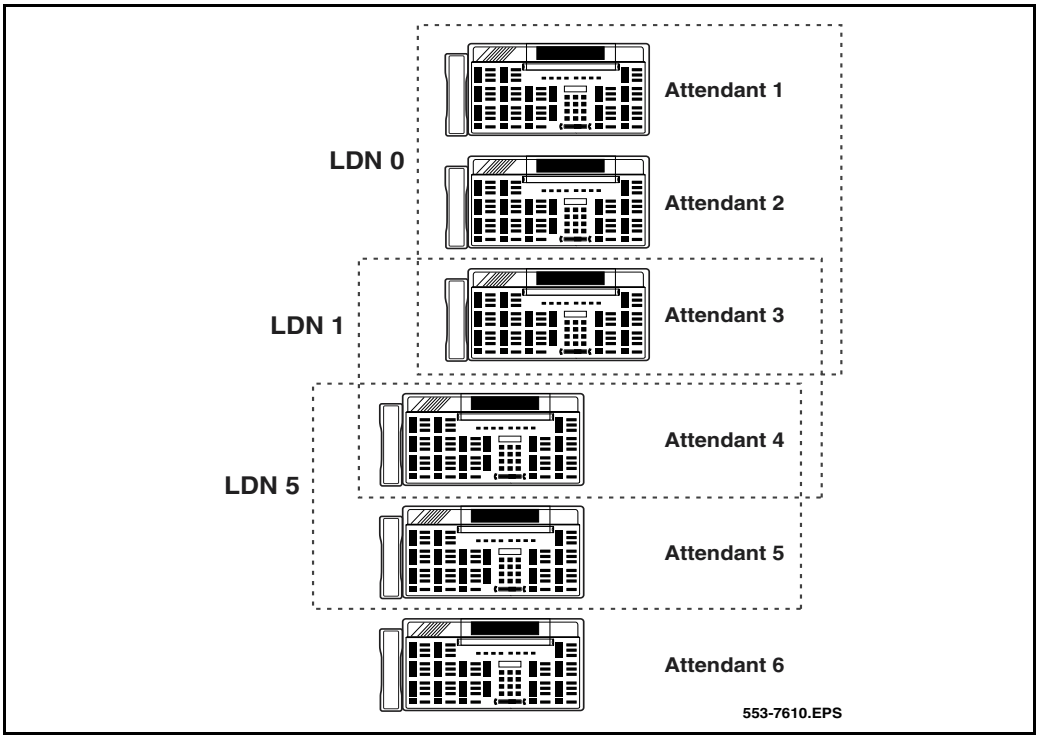
Attendant Emergency Codes

The Attendant Emergency Codes functionality allows an internal/external set to access a group of attendants by dialing an emergency code. This functionality is an enhancement to the existing Departmental Listed Directory Number (DLDN) feature.

The DLDN feature allows specified telephones that share the same numbering plan to belong to one out of a possible six subgroups in a customer. Each DLDN subgroup is identified by one of the customer's Listed Directory Numbers (LDNs). Each department consists of an LDN (0-5) and an associated list of Attendant Consoles (maximum 63) to which LDN calls are delivered.

Figure 24 provides an example of Attendant Console DLDN groupings. These groups are assigned using the LDA prompt in LD 15. In Figure 24, LDN 0 consists of Attendants 1, 2,and 3; LDN 1 consists of Attendants 3 and 4; and LDN 5 consists of Attendants 4 and 5. Attendant 6 does not belong to a DLDN group.

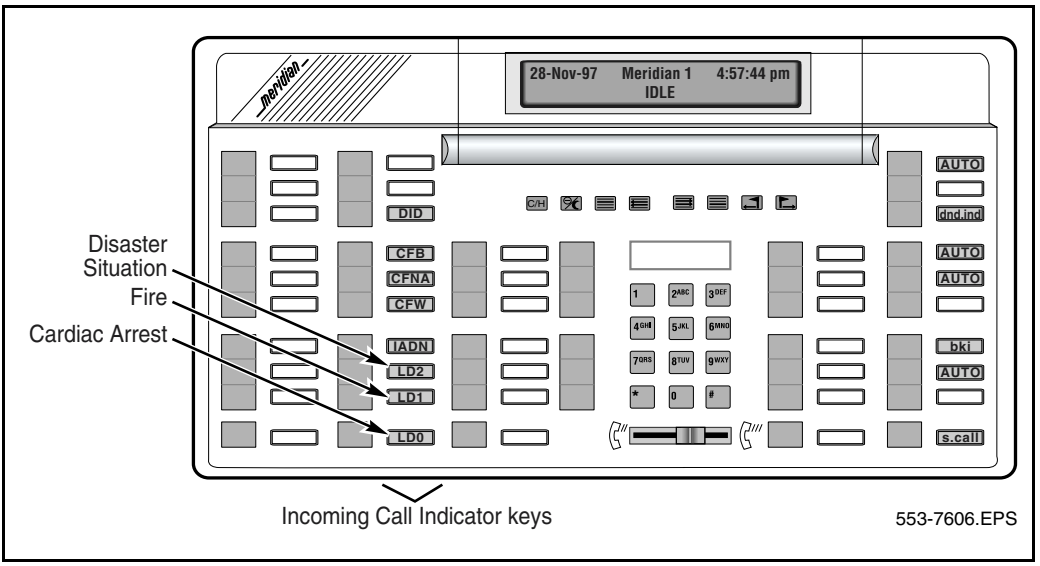
Figure 24
An example of Attendant Console DLDN groupings



ICI keys LDN 0 to LDN 5 can be configured for LDN calls. Emergency code calls use these same ICI keys, as the Attendant Emergency Codes functionality is an enhancement of the DLDN feature. One ICI key can be associated with more than one type of incoming call. Therefore, one ICI key can be configured to answer all emergency code calls.

Figure 25 shows an example of a Meridian 1 Attendant Console with three LDN ICI keys configured. In this example, a hospital has three LDNs that are associated with a particular emergency situation. LDN 0 and the associated ICI key are used for Cardiac Arrest; LDN 1 and the associated ICI key are used for fire emergencies; and LDN 2 and the associated ICI key are used for disaster situations.

Figure 25
Attendant Console with three LDN ICI keys configured



Attendant Emergency Codes functionality provides each DLDN group with the option for Priority Buzzing. Therefore, when the LDN Buzzing (LDBZ) prompt is configured in the Customer Data Block, an audible notification is presented to each of the consoles in the contacted DLDN group. This notification indicates that an emergency code call is waiting to be answered in the attendant queue. The LDBZ prompt allows the selection of each DLDN group that is to be buzzed when an emergency code call is queued.

Therefore, when an internal/external call is placed to LDN 0 as an alert of Cardiac Arrest, all of the attendants in this DLDN group are alerted with Priority Buzzing while the call is waiting in the attendant queue. The LDN 0 ICI key lamp is lit for all Attendant Consoles in the Customer. However, only the attendants of the selected DLDN group receive Priority Buzzing.

The default cadence for Priority Buzzing is two seconds on and ten seconds off. The cadence can be modified by defining the Priority Buzzing cadence (PBUZ) prompt in the Customer Data Block. The Priority Buzzing functionality for emergency code calls is the same as that for IADN calls.

Idle Attendant Console

When an internal/external emergency code call is placed, the system seeks an idle attendant in the DLDN group. The emergency code calls are presented on an idle Loop key in a "Round Robin" fashion. For example, when an LDN call is received, it is presented to the next listed attendant after the one that was last offered a call. This ensures that emergency code calls are distributed in an equitable fashion. Emergency code calls, dial-0 calls, and timed recalls are serviced according to a circular list for the particular LDN.

When the emergency code call is presented on the idle Loop key, the associated ICI key lamp is lit. The ICI lamp status of other Attendant Consoles in the Customer is not updated, since the call is already presented on the Loop key.

Referring to Figure 24, consider the following example:

1 Party 1, an internal set or external trunk, dials LDN 0.

2 The system finds that Attendant Consoles belonging to this group (Consoles 1, 2, and 3) have presentation status for this call.

3 An analysis is now performed to find the attendant that was last offered an LDN call for this group. It is found that Attendant Console 3 was offered the last LDN call.

4 The system attempts to present this call to the next available attendant of this group.

5 The scanning begins with Attendant 2. If Attendant Console 2 is idle, the call is presented to it.

6 If Attendant Console 2 is not available, the system searches for the next Attendant Console (Console 1) in a round robin fashion.

7 When the call is presented to the idle attendant on an idle Loop key, the associated Loop key lamp is lit. Also, the LDN 0 ICI key, if configured, is lit on this console. The Source (SRC) key winks.

8 Once the call is answered, the SRC lamp is steadily lit, and the status of the other lamps remain the same.

When a call is presented to the Attendant Console, the Attendant Console is buzzed continuously by the system, hence Priority Buzzing is **not** applied.

Active Attendant Console

An Attendant Console is in an active state when the Release lamp is dark and the Position Busy key is not activated. When an internal/external call is placed to a DLDN group in which all attendants are active, the call is placed in the attendant queue. The LDN ICI key corresponding to this LDN is updated whenever an Attendant Console of the Customer becomes idle. All active digital Attendant Consoles in the DLDN group receive Priority Buzzing if the LDN ICI key is configured and if this particular DLDN group is defined at the LDBZ prompt.

Referring to Figure 24, consider the following example:

1 Party 1 (an internal set or external trunk) dials LDN0.

2 The system finds that the Attendant Consoles belonging to this group (Consoles 1, 2, and 3) have presentation status for this call.

3 An analysis is now performed to find the attendant that was last offered an LDN call for this group. It is found that Attendant 3 was offered the last LDN call.

4 The system now attempts to terminate the call to the next available attendant of the group.

5 The scanning begins with Attendant 2, the next attendant, and proceeds in a “Round Robin” fashion until an idle Attendant Console is found.

6 If none of the LDN0 consoles are idle, the call is placed in the attendant queue. The call then waits for an idle attendant that has presentation status for the call.

7 The system searches for whether or not the LDN 0 ICI key is configured for the customer. If it is configured, the LDN 0 ICI key lamp is lit for all other Attendant Consoles not in the DLDN group.

8 Priority Buzzing is provided to the digital consoles of this DLDN group, depending on the value of the LDBZ prompt and the configuration of the ICI keys in the Customer Data Block.

When an emergency code call is placed, the corresponding LDN ICI key configuration and the value of the LDBZ prompt is checked. If the DLDN group is included for LDN Buzzing (LDBZ) and if an LDN ICI key is configured, all attendants of the group receive Priority Buzzing.

Referring to Figure 24, consider the following example regarding an active Attendant Console:

1 Party 1 (an internal set or external trunk) dials LDN0.

2 The LDN 0 ICI key lamp is lit for all Attendant Consoles not in the DLDN group.

3 The LDBZ prompt in the Customer Data Block is checked for whether or not LDN0 should be buzzed when an emergency code call is waiting in the attendant queue.

4 When LDN0 is included at the LDBZ prompt, Priority Buzzing is provided to all active digital consoles in this group.

Attendant Consoles 1 and 2 are found to be active and Console 3 in Position Busy. Hence, Consoles 1 and 2 (digital consoles) receive Priority Buzzing.

If Console 3 leaves the Position Busy state, it is presented with the next call in the attendant queue. When the attendant answers the call, Priority Buzzing is provided to the Attendant Console if there is at least one emergency code call waiting in the attendant queue.

5 When a call is waiting in the attendant queue, any one of the attendants in the Customer can pick up the call by pressing the ICI key.

6 When one of the attendants belonging to LDN 0 become free, the first call is presented on an idle Loop key.

7 When the emergency code call is presented, the associated Loop key lamp is lit and the Source (SRC) key lamp winks. Priority Buzzing stops for all of the DLDN attendants of this group and normal continuous buzzing is provided to the console where the call is presented.

8 Once the call is answered, the SRC lamp is steadily lit, and the status of the other lamps remain the same.

If the LDN ICI key is configured and the LDN group is **not** defined at the LDBZ prompt, the Attendant Consoles of the LDN group are **not** provided with Priority Buzzing.

Referring to Figure 24, LDN 0 is not included at the LDBZ prompt; therefore, no buzzing is provided to the LDN0 group of Attendant Consoles. The LDN 0 ICI key lamp is lit for all of the attendants not in the DLDN group.

If the LDN ICI key is **not** configured, the LDN call **does not** receive Attendant Emergency Codes treatment, regardless of how the LDBZ prompt is defined in the Customer Data Block. Without the LDN ICI key configured, the call cannot be taken "out of turn" from the attendant queue, and no Priority Buzzing is provided.

Attendant Console in Position Busy

If all attendants in the DLDN group are in Position Busy when an emergency code call enters the attendant queue, the call is given the same treatment as an LDN call under the same conditions. Since all Attendant Consoles in the LDN group are in Position Busy when the call enters the attendant queue, no Priority Buzzing is provided, and the call remains in the attendant queue. This call updates the corresponding LDN ICI key (if configured) on all available attendants in the Customer. Hence, the attendant can answer the call by pressing the ICI key.

When an Attendant Console leaves the Position Busy mode, all of the emergency code/IADN calls waiting in the attendant queue for this particular attendant receive priority treatment. Hence, if any of the Attendant Consoles in the DLDN group leave the Position Busy state before the call is removed from the queue, Priority Buzzing is provided (if configured).

Customer/Tenant in Night Mode

The customer/tenant is in Night Mode if all of its attendants are in the Position Busy. When an LDN/emergency code call is placed, the call receives the standard night treatment as defined for the customer. If Network Attendant Service (NAS) is equipped and also has NLDN, Priority Buzzing is provided (if configured).

Operating parameters

Existing limitations apply to the Meridian 1 Attendant Console Enhancements feature.

Attendant Console Autoline

Autoline functionality is supported on all Attendant Console types.

Any changes to the Autoline Directory Number must be made in LD 12 and cannot be done on the Attendant Console itself.

The DN programmed on the Autoline key is not verified for validity during configuration. If the DN is invalid, the attendant receives an overflow tone when the Autoline key is used.

The Attendant Autoline key lamp always remains dark.

Individual Attendant Console Directory Number (IADN)

IADN functionality is supported on digital Attendant Consoles (M2250) only.

IADNs must be unique DNs. Therefore, they cannot be Multiple Appearance DNs.

The IADN is a way to contact an attendant and not a DN key. Hence, when an attendant originates a call, the IADN is not relevant.

The IADN can be programmed from the existing range of DID numbers purchased by the customer.

The Calling Party Name Display (CPND) associated with an IADN is the same as the CPND associated with the Attendant DN.

When an IADN call is placed to a particular attendant when the customer/ tenant is in Night Mode, the call receives standard Night treatment as defined for the Customer. During Night Treatment, the call has no priority over other calls in the queue.

As per existing operation, when an attendant places an IADN/LDN call on hold and the system initializes, the IADN/LDN call that is on hold is lost.

As per existing operation, when there is an IADN/LDN call in the attendant queue and the system initializes, all calls in the queue are dropped.

The Call Waiting lamp on the Attendant Console reflects the IADN calls waiting in the attendant queue.

When an IADN call is placed in the attendant queue, a maximum of a two second delay may occur before Priority Buzzing begins.

When an Attendant Console is service changed in LD 12 while it is active, the Attendant Console goes into a Position Busy state. In this case, Priority Buzzing stops for any buzzing IADN call waiting in the attendant queue.

If the Attendant Console is service changed in LD 12 and REQ = OUT, all IADN calls to this attendant are treated as normal attendant calls and are presented to any available attendant in the Customer/Console Presentation Group. ICI keys are not lit for these calls on other Attendant Consoles.

If the Attendant Console is service changed in LD 12 and REQ = CHG, all IADN calls for this attendant are presented to any available attendant in the Customer/Console Presentation Group, unless the IADN attendant leaves the Position Busy state before the call is taken out of the attendant queue and the attendant number is not changed. If this is the case, the call receives priority treatment as defined for IADN.

During service change, when the IADN DN is changed, the IADN calls for the originally intended Attendant Console can still terminate to that console as long as the attendant number remains the same.

When the IADN ICI key configuration is removed from the Customer Data Block, or if IDBZ = NO, then Priority Buzzing for any IADN calls waiting in the attendant queue is stopped. The attendant is no longer able to answer the IADN call “out of turn” from the attendant queue if the ICI key is removed.

If the IDBZ prompt is changed from NO to YES, the IADN calls waiting in the attendant queue do not apply Priority Buzzing to the respective attendants. However, when a new IADN call is placed in the attendant queue, the Attendant Console receives Priority Buzzing within two seconds for all of the IADN calls waiting for this particular console in the attendant queue.

If an IADN ICI key is configured for the Customer, Priority Buzzing is not provided for the IADN calls that are already waiting in the attendant queue. Priority Buzzing is only provided when new IADN calls are placed in the queue.

If the IADN ICI key is not configured, the IDBZ prompt is still given, but its value is ignored. Therefore, Priority Buzzing is not provided in this case.

When an IADN call is placed in the attendant queue and waits for a console that is already being buzzed (Recall Buzzing, Attendant Emergency Codes Priority Buzzing, or another IADN call Priority Buzzing) Priority Buzzing is not provided immediately.

Priority Buzzing is not provided when an IADN call is presented to an idle Attendant Console with normal buzzing.

IADNs cannot be configured as an Attendant Alternative Answering (AAA) DN, Attendant Overflow Position (AOP) DN, or Night DN.

Data calls to an IADN are not supported.

An IADN can be configured as a valid intercept computer DN.

An attendant cannot place a call to another attendant on the same node by dialing the attendant's IADN. If an attendant tries to do this, an overflow tone is given.

Attendant Emergency Codes

All existing limitations/interactions of the DLDN feature apply to emergency code calls.

Attendant Emergency Codes functionality is supported on digital Attendant Consoles (M2250) only.

The DLDN package must be equipped and enabled in order for Attendant Emergency Codes to function.

Attendant Emergency Codes functionality is supported at a customer level only.

A DLDN group may contain any type of Attendant Console; however, only digital consoles receive Priority Buzzing.

When the Attendant DN 0 is called, the call is routed to only those Attendant Consoles belonging to the LDN group. Dial 0 and Slow Answer Recalls are not treated as emergency code calls, and no Priority Buzzing is provided, regardless of how the LDBZ prompt is defined.

When an emergency code call is placed in the attendant queue, a maximum of a two second delay may occur before Priority Buzzing begins.

Each DLDN that is configured as an emergency code number decreases one customer LDN.

When an Attendant Console is service changed in LD 12 while it is active, the Attendant Console goes into a Position Busy state. In this case, Priority Buzzing stops for any buzzing emergency code call waiting in the attendant queue. If the Attendant Console leaves the Position Busy state while the emergency code call is still waiting in the attendant queue, the console receives Priority Buzzing.

If a new LDN ICI key is configured for the Customer, Priority Buzzing is not provided for the emergency code calls that are already waiting in the attendant queue. The console receives Priority Buzzing for new emergency code calls placed in the attendant queue.

If a new DLDN group is defined at the LDBZ prompt, Priority Buzzing is not provided for the emergency code calls that were already waiting in the attendant queue. When a new emergency code call is inserted in the attendant queue, however, Priority Buzzing is provided if the corresponding LDN ICI key is configured.

If an Attendant Console is removed from its LDN group while an emergency code call is waiting in the attendant queue, Priority Buzzing is stopped. The status of the LDN ICI key lamp remains the same. Also, the Attendant Console loses its presentation status.

If the ICI key and the LDBZ/IDBZ prompts are not configured appropriately, there may be calls waiting in the attendant queue that are not providing Priority Buzzing to any consoles. If, through service change, the ICI key and the IDBZ/LDBZ prompt are then configured appropriately, Priority Buzzing is still not provided until another valid call enters the queue or the appropriate attendant enters the Position Busy state and then leaves the Position Busy state.

When an Attendant Console is added to an LDN group, Priority Buzzing is not provided to the console for the emergency code calls that are already waiting in the attendant queue. The console only receives Priority Buzzing for new emergency code calls inserted in the attendant queue.

If the LDN ICI key configuration is removed from the Customer Data Block, or if an LDN group is removed from the LDBZ prompt, then Priority Buzzing for any emergency code calls waiting in the attendant queue is stopped. The attendant will no longer be able to answer the DLDN call “out of turn” from the attendant queue.

An emergency code call that enters the attendant queue to wait for an Attendant Console which is already being buzzed (for example, the recall buzzer, IADN Priority Buzzing, another Attendant Emergency Codes Priority Buzzing) is not given priority treatment immediately.

When an emergency code call is waiting in the attendant queue, the ICI key lamp on the Attendant Console is the only visual indication of the emergency. Audible indication, Priority Buzzing can still be provided.

Priority Buzzing is not provided when an emergency code call is presented to an idle Attendant Console with normal buzzing.

Feature interactions

Attendant Console Autoline

The feature interactions for Attendant Console Autoline are similar to those for Attendant Autodial.

Individual Attendant Console Directory Number (IADN)

Attendant Console

The Attendant Console feature provides equal load distribution among all available attendants. When an IADN call has been handled by an attendant, the system does not consider this attendant as the attendant last used.

Attendant Emergency Codes

If an attendant is already being buzzed for an emergency code call and an IADN call is placed in the attendant queue to wait for this particular attendant, Priority Buzzing is not provided immediately for the IADN call.

Attendant Alternative Answering

Presented IADN calls are given Attendant Alternative Answering (AAA) treatment as defined for the customer. After the predefined timing threshold, unanswered IADN calls are forwarded to the AAA DN. The AAA DN of a console cannot be defined as an IADN.

Attendant Calls Waiting Indication

The Call Waiting lamp on the console winks when the Call Waiting queue Update (CWUP) prompt is set to NO and there is at least one IADN call waiting in the attendant queue for the particular console.

If CWUP = YES in the Customer Data Block, the Call Waiting count on the console includes the IADN calls waiting in the queue. When CWUP = YES, the Call Waiting lamp always remains lit.

If a console is in Position Busy, the IADN call is counted against the Console Presentation Group (CPG) and it is reflected on all consoles of that particular CPG.

Attendant Forward No Answer

IADN calls are given Attendant Forward No Answer (AFNA) treatment as defined for the customer. If an IADN call is not answered in the specified time, it is put back in the attendant queue and the console is put in Position Busy mode. The IADN call now loses its priority and can terminate to any of the available Attendant Consoles or the NITE DN.

Attendant Overflow Position (AOP)

An IADN call is not forwarded to the Attendant Overflow Position (AOP) DN as long as the intended attendant is available. This is because the addressed attendant is still available and the call can eventually terminate to it once it is placed in the queue.

If the attendant is in Position Busy, its IADN calls will be forwarded to the AOP DN. The AOP DN cannot be an IADN.

Attendant Recall (Slow Answer Recall)

The Slow Answer Recall feature is not affected by Meridian 1 Attendant Console Enhancements.

For call presentation, slow answer recalls take priority over all other calls in the attendant queue. When an active attendant becomes idle, the system first searches for any recalls waiting to be presented and then it attempts to present calls from the main attendant queue.

For an IADN call to be recalled to the same attendant, the Recall to Same Attendant (RTSA) feature must be configured. The Recall ICI key lamp is lit when an IADN call slow answer recalls back to the attendant. The IADN ICI key lamp is not lit in this case, and Priority Buzzing is not applied.

When an analog (500/2500 type) set transfers a call to an IADN, this call is treated as an IADN call whether it is in the queue or presented to the console. When this call is presented to the console, the ICI key lamp is lit and the call is split onto the source and destination sides, as per existing recall functionality.

Attendant Recall (Set Recall)

For call presentation, set recalls do not take priority in the attendant queue.

When an analog (500/2500 type) set without CLS = XFA/TSA performs a switch hook flash or when a Meridian 1 proprietary set presses the Attendant Recall (ARC) key during an established call, this call is treated as an attendant recall. The Recall ICI key lamp is lit and the dialed DN is shown as the Attendant DN.

When an analog (500/2500 type) set with CLS = XFA/TSA performs a switch hook flash and then dials an IADN, the call is treated as a regular set recall while in the attendant queue. The Recall ICI lamp is lit while in the attendant queue. Once this call is presented to the console, it is split onto the source and destination sides as a recall normally does. The IADN ICI lamp is lit.

When a proprietary set transfers a call to an IADN, Set Recall functionality is applicable.

Automatic Call Distribution

An IADN can be configured as an ACD Night DN. When an ACD Night call attempts to terminate on the Attendant Console, it is treated as a priority call for this attendant.

Console Operations

The IADN feature overrides the presentation status defined by the Console Operations (COOP) feature. Therefore, even if presentation status is denied on the IADN ICI key, IADN calls are automatically presented on the Loop key.

Call Redirection features

Whenever an IADN call is made as a result of Call Redirection, this call receives the standard IADN treatment (that is, Priority Buzzing and IADN ICI). The Attendant IADN feature does not distinguish between forwarded calls and direct dial IADN calls.

Hunt

If an IADN is defined as part of a Hunt chain, calls terminate to the IADN, following the Hunt chain. Once a call is placed in the attendant queue, however, the next DN in the Hunt chain is not sought.

If the IADN console is in Position Busy, the call is presented to any one of the available attendants in the Customer/Console Presentation Group. Therefore, the next DN in the Hunt chain is not sought once an attempt is made to present the call to the IADN attendant.

Message Center

If an IADN is given as an MWK DN, the Message Waiting calls receive IADN treatment. Therefore, Priority Buzzing is provided, and the IADN ICI key is lit (if configured).

Multi-Tenant Service

Sets belonging to a Customer can be divided into customer subgroups known as tenants. A set belonging to one tenant can call an attendant belonging to another tenant by dialing the attendant's IADN. The IADN functionality takes precedence over the Multi-Tenant Service feature.

Network Attendant Service

Network Attendant Service (NAS) treatment is applied when the Customer/ Console Presentation Group is in Night mode. An IADN call rerouted using NAS loses its priority at the remote node.

If the NAS ID of one node is defined as the IADN/emergency code number of the remote node, priority treatment is provided to all redirected calls, including IADN/emergency code calls. In this case, the IADN ICI key has a higher precedence than the corresponding NAS ICI key.

Network Message Services

The IADN ICI key takes priority over the MWC ICI key. When a call is forwarded to an IADN over a network to a Message Center, the call receives Priority Buzzing and the IADN ICI key is updated (if configured).

Night Service

When the system is in Night mode and an IADN call is the next call to be presented, the call receives Night treatment as defined for the Customer. Priority Buzzing is not provided to the Night DN. The IADN call is presented to the Night DN whenever its "turn" comes about.

If the system returns to Day Mode, the remaining IADN calls in the attendant queue are provided with priority treatment. The Night DN for the Customer/ Console Presentation Group cannot be an IADN.

Enhanced Night Service

IADN calls from the public network lose their priority treatment when presented to the Enhanced Night DN. If the system returns to Day Mode, the remaining IADN calls in the attendant queue receive priority treatment. The Enhanced Night DN of a trunk cannot be an IADN.

Permanent Hold

If a set is in Permanent Hold and dials an IADN, the set receives overflow tone.

Attendant Emergency Codes

Attendant Forward No Answer

When an unanswered emergency code call is given Attendant Forward No Answer (AFNA) treatment, it is placed back in the attendant queue when it is not answered within the specified time, and the console is placed in Position Busy. If the other consoles of this particular DLDN group are in an active state, Priority Buzzing is provided for them, depending upon the configuration of the LDN ICI key and the value of the LDBZ prompt.

Individual Attendant Directory Number

If an Attendant Console is already receiving Priority Buzzing for an IADN call that is waiting in the attendant queue, Priority Buzzing is not provided immediately for an emergency code call that enters the attendant queue.

Network-wide Listed Directory Number

When the DLDN dialed at one node is configured as an emergency code number at a remote node, a call routed using Network Attendant Service (NAS) (when Network-wide Listed Directory Number (NLDN) is configured) terminates at the remote node and receives priority treatment.

Feature packaging

The Attendant Console Autoline and the Individual Attendant Console Directory Number (IADN) functionalities are included in base System Software. For Attendant Emergency Codes functionality, however, Departmental Listed Directory Number (DLDN) package 76 is required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure Priority Buzzing and an Individual Attendant Directory Number (IADN) Incoming Call Indicator (ICI) key for a digital Attendant Console.

2 LD 15 – Configure Departmental Listed Directory Number (DLDN) and Priority Buzzing for Attendant Emergency Code calls.

3 LD 12 – Configure an Autoline DN for an Attendant Console.

4 LD 12 – Configure an Individual Attendant Directory Number (IADN) for a digital Attendant Console.

LD 15 – Configure Priority Buzzing and an Individual Attendant Directory Number (IADN) Incoming Call Indicator (ICI) key for a digital Attendant Console.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	2250 1250	Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

IDBZ	YES	Individual Attendant DN Buzzing-on for IADN calls in the attendant queue. NO = Individual Attendant DN Buzzing-off for IADN calls in the attendant queue (default).
PBUZ	xx yy	Flexible Priority Buzzing cadence for IADN and Attendant Emergency Code calls, where: xx = Priority Buzzing - on phase yy = Priority Buzzing - off phase The PBUZ range is from 2 to 16 seconds. If the value entered is an odd number between 2 and 16, it is rounded down to the next lowest even integer.
...		
ICI	xx IADN	ICI key for individual Attendant DN, where: xx = ICI key number (0 - 19).

LD 15 – Configure Departmental Listed Directory Number (DLDN) and Priority Buzzing for Attendant Emergency Code calls.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	LDN	Listed Directory Numbers.
CUST	xx	Customer number, as defined in LD 15
DLDN	YES	Departmental Listed Directory Numbers.
...		
LDN5	xxxx	Emergency code number.
LDA5	1-63	M2250 Attendant Console associated with LDN5.

ICI	xx LD0 xx LD1 xx LD2 xx LD3 xx LD4 xx LD5	Incoming Call Indication for Listed Directory Numbers 0-5. xx = key number 00-19.
LDBZ	n n n n n n	The DLDN groups which should be buzzed when an LDN/ emergency code call is in the attendant queue, where: n = 0, 1, 2, 3, 4, and/or 5.

LD 12 – Configure an Autoline DN for an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250 1250	Attendant Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
KEY	nn AUTO xxx...x	Direct Autoline DN, where: nn = Key number (0 - 19) and xxx....x = Autoline DN. The Autoline DN can be 1-31 digits in length.

LD 12 – Configure an Individual Attendant Directory Number (IADN) for a digital Attendant Console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

ANUM ... IADN	1 - 63 xxxx	Attendant Number. Individual Attendant DN for this Attendant Console. The Individual Attendant DN can be 1-4 digits in length or 1-7 digits in length if DNXP package 150 is equipped. The IADN cannot be a Multiple Appearance DN.

Feature operation

Attendant Console Autoline key

To place an Autoline call:

1 The attendant presses a Loop key. The Loop key lamp is lit.

2 The attendant presses the Autoline key. The pre-programmed number on the Auto-line key is automatically dialed. The Source (SRC) lamp on the Attendant Loop key winks.

3 The dialed party answers the call, and the SRC key lamp is steadily lit.

To extend a currently active call to the Autoline DN:

1 The attendant is active on an established call.

2 The attendant presses the Autoline key to extend the call.

3 The pre-programmed number on the Autoline key is automatically dialed. The destination (DEST) lamp on the Attendant Console winks.

4 The dialed party answers the call, and the DEST lamp is steadily lit.

5 To complete the transfer, the attendant presses the Release (RLS) key. Once the Release key is pressed, the display is cleared.

To display the DN programmed for the Autoline key, the attendant presses the Autoline key when the console is idle or in Position Busy.

On an analog console, to display a DN that is longer than eight digits, the attendant presses the display key after pressing the Autoline key.

Individual Attendant Directory Number

The following is an example of Individual Attendant Directory Number (IADN) functionality for an active Attendant Console with an IADN ICI key configured. Also, the IDBZ prompt set to YES in the Customer Data Block.

1 An attendant is involved in an active call.

2 An IADN call is placed to the active attendant and waits to be answered in the attendant queue.

3 Priority Buzzing is provided to the Attendant Console. During this time, if another IADN call for the same attendant, is placed in the attendant queue, the Priority Buzzing is not affected.

4 The attendant releases the active call.

5 The next call in the queue is presented to the attendant.

6 The Priority Buzzing stops, and the attendant receives a continuous buzz for the newly presented call. All ICI keys on the Attendant Console, including the IADN key, are updated. The IADN ICI key lamp flashes if there is at least one IADN call waiting in the attendant queue.

7 The attendant chooses to answer the IADN call, from the queue, by pressing the IADN ICI key. If there is another IADN call waiting for the attendant in the queue, Priority Buzzing is applied to the attendant again. If there is not another IADN call waiting, then the Priority Buzzing stops. If the attendant selects another call over the IADN call (using another ICI key or taking a non-IADN call if presented on the Loop key), Priority Buzzing begins again.

Attendant Emergency Codes

The following is an example of Attendant Emergency Codes functionality for Attendant Consoles with an LDN ICI key configured. Also, the DLDN group is included for LDN Buzzing at the LDBZ prompt. Referring to Figure 24:

1 Party 1 (an internal set or external trunk) dials LDN0.

2 The LD0 ICI key lamp is lit for all Attendant Consoles not in the DLDN group.

3 The LDBZ prompt in the Customer Data Block is checked for whether or not LDN0 should be buzzed when an emergency code call is waiting in the attendant queue.

4 LDN0 is included at the LDBZ prompt. Therefore, Priority Buzzing is provided to all active digital consoles in this group.

Attendant Consoles 1 and 2 are found to be active and Console 3 in Position Busy. Hence, Consoles 1 and 2 (digital consoles) receive Priority Buzzing.

If Console 3 leaves the Position Busy state, it is presented with the next call in the attendant queue. When the attendant answers the call, Priority Buzzing is provided to the Attendant Console if there is at least one emergency code call still waiting in the attendant queue.

5 When a call is waiting in the attendant queue, any one of the attendants in the Customer can pick up the call by pressing the ICI key.

6 When one of the attendants belonging to LDN0 become free, the first call is presented on an idle Loop key.

7 When the emergency code call is presented, the associated Loop key lamp is lit and the Source (SRC) key lamp winks. Priority Buzzing stops for all of the DLDN attendants of this group and normal continuous buzzing is provided to the console where the call is presented.

8 Once the call is answered, the SRC lamp is steadily lit, and the status of the other lamps remain the same.

Meridian 1 Initialization Prevention and Recovery

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This section contains information on the following topics:

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Feature description

The Meridian 1 Initialization Prevention and Recovery feature reduces the occurrences of initializations by tracking specific hardware faults and automatically disabling the affected hardware locally. This feature offers the following specific functionalities:

- Network Loop Response Time-out Initialization Prevention (LRIP)
- Serial Data Interface Device Response Time-out Initialization Prevention (SRIP)
- Network Loop Overload Initialization Prevention (LOIP), and
- Localized Faulty Hardware Recovery (FHWR).

Network Loop Response Time-out Initialization, Serial Data Interface Device Response Time-out Initialization Prevention, and Network Loop Overload Initialization Prevention are designed to prevent system initialization. The function of Localized Faulty Hardware Recovery is to automatically disable any faulty loops, Serial Data Interface (SDI) devices or Expanded Serial Data Interface (ESDI) devices identified by this feature.

Network Loop Response Time-out Initialization (LRIP)

When a network loop fails to respond to a processing request, the LRIP function is automatically invoked to avert a system initialization. An FHW000 message is printed on all maintenance TTYs to notify the system administrator of the faulty loop. The loop is marked as faulty in the system database.

Serial Data Interface Device Response Time-out Initialization Prevention (SRIP)

When a Serial Data Interface (SDI) or ESDI device fails to respond to a processing request, the SRIP function is automatically invoked to avert a system initialization. An FHW001 message is printed on all maintenance TTYs to notify the system administrator of the faulty SDI. An FHW002 message is printed on all maintenance TTYs to notify the system administrator of the faulty ESDI. The device is marked as faulty in the system database.

Network Loop Overload Initialization Prevention (LOIP)

When loop overload is detected, the LOIP function is automatically invoked to avert a system initialization. This function disables the signaling capability of the network loop and marks it as faulty in the system database before allowing the existing processing to continue. An FHW003 message is printed on all maintenance TTYs to indicate the faulty network loop and to indicate that an INI000 0006 has been averted. The device is marked as faulty in the system database.

Localized Faulty Hardware Recovery (FHWR)

Once a network loop, SDI or ESDI is identified as being faulty, it is tracked by the FHWR function. When the system is available to load and run a background routine and the faulty network loop, SDI or ESDI device is still in enabled status, an appropriate maintenance overlay is automatically invoked to disable it. A technician can also manually disable it by using existing maintenance overlay commands. The faulty loop, SDI or ESDI device is tracked by the FHWR function until the loop is disabled.

When a maintenance overlay is running and Multi-User Login is not enabled, an OVL111 xx FHWR message is given prior to a user logging into the system to indicate that the system is automatically performing the FHWR maintenance task. If the user does log in, the FHWR maintenance task is interrupted; when the user logs out, the FHWR function will reload the maintenance overlay to resume disabling the faulty hardware. Once it has disabled the loop, an FHW004 message is printed on all maintenance TTYs to indicate that a faulty network loop has been automatically disabled and the maintenance overlay has terminated (the message FHW005 is printed for an SDI device and FHW006 for an ESDI device). The device is marked as faulty in the system database.

Operating parameters

This feature applies to Large Systems.

After the Network Loop Overload Initialization Prevention function has identified a faulty network loop, if there are trunks configured on the hardware, far-end seizure of such trunks are treated in the same manner as a non-responding trunk.

Feature interactions

Meridian 1 Fault Management

FHW000, FHW001, FHW002, FHW003, FHW004, FHW005, and FHW006 can be defined as a trigger string that is monitored by the Meridian 1 Fault Management feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Meridian 911

Contents

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Feature description

The number 911 has been adopted for the purpose of reporting emergencies and requesting emergency services. For localities with 911 systems, the number:

- is the same in all communities
- is easily remembered, even under adverse conditions
- provides direct telephone access to emergency services regardless of the time of day, or the caller's familiarity with an area, or the caller's ability to identify the type of emergency

A 911 system is planned, implemented, and operated under the auspices of local governments. In most communities, 911 provides access to police, fire, and emergency medical services. In some locations additional services are accessible (for example, dialing 911 in certain locations provides access to Coast Guard search and rescue services). Approximately 80 percent of all 911 calls are intended for the police, with the balance split between fire and ambulance.

Because the overwhelming majority of 911 calls require police attention, local police departments generally maintain, manage, and staff the center to which emergency calls are first directed. These centers are referred to as primary answering centers. A secondary answering center could be a police, fire, or ambulance station (for example, fire-related 911 calls may be transferred to a secondary answering center that handles incoming calls regarding fires). In many instances, the fire department also determines the degree of urgency for emergency medical services.

If the primary or secondary answering center is busy or out of service, the 911 call is directed to a backup answering center, referred to as an alternate answering center.

The public network routes a 911 call to the appropriate primary answering center based on the caller's telephone number. For this reason, callers dialing 911 give up their right to privacy regarding:

- the telephone number of the station from which they are calling, and
- the billing address associated with that telephone number.

To protect a caller's right to privacy, some communities still allow the use of seven-digit emergency numbers, routed either to an answering center or directly to the responding agency.

Basic 911 service

Basic 911 service routes emergency calls to an answering center based on the location of the Public Exchange/Central Office serving the calling station. The jurisdiction of an answering center is determined by the Central Office boundaries. The most basic 911 system involves only one Central Office and one exchange service area, and can be a single answering center.

Enhanced 911 service

In areas where telephone company Central Office boundaries do not match jurisdictional boundaries, there is a problem in identifying which emergency agency should receive the emergency call. There may be an even more complicated situation if the 911 network includes two or more primary answering centers, and each serves areas that do not match the Central Office serving areas.

Enhanced 911 (E911) service ensures that an emergency call originating in any particular jurisdiction covered by the 911 system is recognized and forwarded to the appropriate responding agency in the same political or geographical jurisdiction as the originating call.

Enhanced 911 service uses more sophisticated equipment and features than basic 911 service. Specialized features include:

- Automatic Number Identification (ANI)
- Automatic Location Identifier (ALI), and
- Selective Routing (SR).

Display of the ANI associated with the originating call sometimes replaces the need for the following basic 911 options: Called Party Hold; Emergency Ringback; and Switchhook Status. Therefore, sometimes these features are not provided with enhanced 911 service.

The Automatic Number Identification (ANI) of a 911 call consists of eight digits (a Numbering Plan or Information digit followed by the seven digits of the calling party number). Whether the first digit of the ANI string is to be interpreted as a Numbering Plan Digit (NPD) or an Information Digit (ID) depends on the trunk interface and Meridian 911 configuration.

> ***Note:*** The 10/20 digit ANI on 911 calls feature brings the system into compliance with the Federal Communications Commission (FCC) decision that requires a circuit switched network, working as a Public Safety Answering Point (PSAP), to accept a 10 or 20 digit ANI when terminating 911 calls. For more information on the 10/20 Digit ANI on 911 Calls feature, please refer to the "10/20 Digit ANI on 911 Calls" feature description in *Features and Services* (553-3001-306), Book 3.

The Automatic Location Identifier (ALI) host computer uses the ANI to locate the ALI record for the calling party number. This includes the name and address, and whether the line is business or residence. An enhanced 911 system creates ALI information from the ALI record and automatically routes the ALI information to an optional data terminal display at the answering center.

An enhanced 911 system routes all emergency calls from the originating Central Offices through an E911 Tandem, sometimes called a 911 control office, to the primary answering center. There, using Selective Routing features, a call taker can transfer the call through the public network by signaling the E911 Tandem. The Autodial Tandem transfer feature can be used for this. For example, if the primary answering center transfers calls to several fire departments, it uses one fire department button. The option automatically:

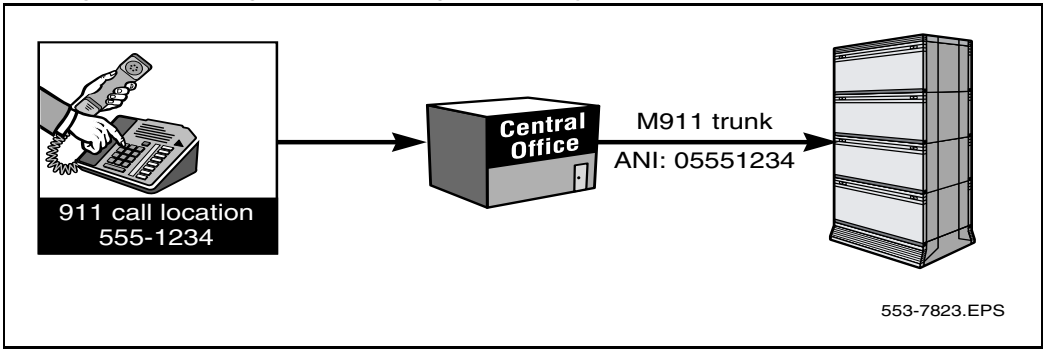
- identifies the fire department associated with the caller's location, and
- transfers the call to that department.

Meridian 911 (M911) system

The Meridian 911 system:

- gives priority to emergency calls
- routes priority calls, without interrupting service, to answering positions that can identify and dispatch the assistance required with minimum delay
- displays the calling party's number
- puts the calling party number into Call Detail Recording (CDR) Q and N records, and
- provides an external notification that an emergency call is queued.

Figure 26
Routing the call, along with the ANI digits, to the system



When a call arrives at the system through an M911 trunk, the trunk software in the system communicates with the serving Central Office (CO) (either the local Central Office or the M911 tandem office) to receive the ANI information through multifrequency (MF) signaling. When all ANI digits are received, the system software starts to process the call.

Meridian 911 Call Abandon

A 911 call is considered abandoned by the system if the call terminates on a 911 trunk route, and the calling party disconnects after trunk seizure, but before the call is answered. This can occur while the call is waiting in an Automatic Call Distribution (ACD) or Controlled DN (CDN) queue, or when the call is presented to the ACD agent but is not yet answered.

The Call Abandon feature allows the system to treat an abandoned call as though the calling party is still connected. The call maintains its place in the ACD queue, and is presented to an agent. When the agent answers, the agent receives a continuous, cadenced six-second tone, as well as an indication on the set's display, to indicate that the call is an abandoned call. Automatic Number Identification (ANI) information is also displayed. The agent can then call back the originator of the call.

Once the call is abandoned, the trunk is released for other 911 calls. Information on abandoned calls can be included in Call Detail Recording (CDR) records if New Format CDR (FCDR) package 234 is equipped.

Operating parameters

Meridian 911

Meridian 911 routes are restricted to incoming traffic only.

Incoming M911 trunks use MF signaling only. Dial Pulse (DP) and Dual-tone Multifrequency (DTMF) are not supported for M911 routes.

911 Calls on Integrated Services Digital Network (ISDN) trunks are not supported.

A call is considered a 911 call by system software if it arrived on a trunk belonging to an M911 route. Calls dialing 911 internally can, through configuration of the Electronic Switched Network (ESN) digit manipulation tables, be terminated locally (for example, to a Controlled DN), but these calls are internal calls to the software, not 911 calls.

ANI is expected for every call. Meridian 911 does not support 911 calls from an E911 Tandem which does not support sending ANI.

The priority of incoming trunk calls internally transferred to an Automatic Call Distribution (ACD) DN queue (a secondary answering center) may be preserved using blind transfer only. All other types of call modification (for example, consultation transfer, or conference) are treated as internal calls and the calls are linked to the low priority queue of the ACD DN.

The No Hold Conference feature, the recommended feature for transferring calls between answering positions, is not available on analog (500/2500 type) telephones.

The Call Prioritization (911 calls presented with higher priority) and Call Waiting Notification features are applicable to ACD answering centers only. These cannot be supported on Multiple Appearance Directory Number (MADN) answering centers.

The first answering center must be an ACD DN.

M911 trunk calls must terminate on a CDN. If an autoterminate DN is specified that is not a CDN, an SCH error message is printed. If a CDN is used as the autoterminate destination of at least one M911 trunk, the CDN cannot be removed using LD 23 (an SCH message will be given). To remove the CDN, all M911 trunks terminating to it must be removed, or they must be changed to terminate to a different CDN.

CDNs as well as ACD DNs are normal dialable numbers. Nothing prevents non-911 calls from arriving at either the CDN, or any of the ACD DNs acting as answering centers using direct dialing. Non-911 calls arriving at CDNs are defaulted to the CDN's default ACD DN; non-911 calls arriving at an ACD DN are treated as normal calls.

The Call Waiting Notification (CWNT) package 225 is a separate package and an M911 system can be installed without it. If the package is not equipped, no external alert can be given for 911 calls arriving at an ACD queue.

The CWNT software is available for 911 calls in ACD queues only. There is no provision for alerting MADN call takers of arriving 911 calls.

911 calls in an ACD queue are treated the same as other ACD calls. Therefore, if Recorded Announcement (RAN) is configured for the ACD queue, 911 calls will be given RAN treatment. The same interactions between RAN and Central Office loopstart trunks exist for M911 as they do for general ACD operation.

Meridian 911 Call Abandon

Calls released by the originator after the call has been answered are not calls abandoned by the definition used for the M911 Call Abandon feature and do not receive abandon treatment.

Abandoned calls waiting in the ACD queue activate the Call Waiting Notification Terminal Number.

If ANI is not received, the abandoned call is not presented to the agent since it is no longer useful; however, a Call Detail Recording (CDR) N record, if configured, can be printed to indicate that the call has abandoned.

Only external 911 calls abandoned before answer are supported.

When the call is abandoned, the speech path is dropped, and the trunk is released.

If Flexible Tones and Cadences (FTC) package 125 is equipped, it is possible to configure a tone other than the one provided by default.

Call Abandon is configured on a per route basis.

Call Abandon is supported on 911 trunks only.

No B record is generated by CDR for an M911 abandoned call, because the B record is package dependent and only applies to an established call with Internal CDR.

Wireless sets are not supported at the Public Safety Answering Point (PSAP) or Secondary Safety Answering Point (SSAP) for Call Abandon.

An MF tone receiver (QPC916 or NTAG20AA) is required.

Feature interactions

10/20 Digit ANI on 911 Calls

The 10 Digit ANI feature changes the ANI format to include the NPA in the ANI field. A single PSAP can handle any number of valid NPAs with the 10 digit format.

The 20 digit ANI feature addresses the problem of accurately determining the location of a wireless calling party dialing 911. The first 10 ANI digits provide the Calling Station Number (CSN). The CSN for a 911 call is the Calling Party Number (CPN), if available, or the billing number if the CPN is not available. The CPN, if available, is used to call the originator back when a 911 call is disconnected.

The second 10 ANI digits, or Pseudo Automatic Number Identification (PANI), provides the cell site and sector information to best define the wireless calling party's location. The PANI allows emergency assistance to be sent to the correct area.

Automatic Call Distribution interactions

ACD-C Reports

The Meridian 911 product does not change the ACD-C reports. M911 will use the ACD-C reports for CDNs as introduced for Customer Controlled Routing (CCR).

Only four of the fields in the report will have any meaning. Because M911 uses the Route-to Application Module Link (AML) message instead of the Queue-to message, only “Route To”, “Default DN”, “Abandoned”, and “Calls Accepted” are meaningful. Those calls that are successfully routed count towards the “Route To” category. Those calls that get default treatment count towards the “Default DN” category. Those calls that abandon while they are in the CDN queue count towards the “Abandoned” category. The “Calls Accepted” category will be the sum of the “Route To”, “Default DN”, and “Abandoned” categories.

The “# of Calls in the Queue” category represents those calls that are sitting in the CDN queue. This should always be zero, since calls waiting for a Route-to request from the Application Module are sitting in a timing queue as opposed to the CDN queue.

M911 calls routed to an ACD answering center will show up in the normal ACD queue and agent reports for that queue. Calls routed to MADN answering centers will show up only in the CDN report.

ACD-D Auxiliary Message

No changes to the ACD-D reports are needed for Meridian 911.

Controlled Directory Number (CDN) Ceiling

The CDN ceiling feature returns busy tone to calls arriving at the CDN while it is in default mode. If a 911 call should arrive while these conditions are true, the 911 call will not hear busy tone, but will be linked into the default destination ACD DN’s queue. Therefore, the setting of the ceiling value is irrelevant if only 911 calls are expected at the CDN. The ceiling value will, however, still be applied to non-911 calls arriving at the CDN.

Controlled Directory Number (CDN) Ringback

911 calls get ringback immediately upon arrival at a CDN, whereas CCR calls do not.

Customer Controlled Routing (CCR) Call Abandoned Message (ICAB)

This message is sent for controlled calls that were abandoned before being answered.

Customer Controlled Routing (CCR) Call Enters Queue Message (ICEQ)

This message is sent to ACD-MAX each time a default call is placed in the default ACD DN (default mode).

Customer Controlled Routing (CCR) Call Modification Message (ICCM)

This message is sent to ACD-MAX when a call modification request (route to, disconnect, busy) is successfully executed upon a CDN controlled call.

Note that since the Route To, Disconnect and Busy treatments remove CDN control from the call, ICCM messages will be sent for the call for each of the queues from where it must be removed. The ICCM message also applies to Enhanced ACD Routing calls or CDN default calls which were busied by the call ceiling value while trying to route to the default ACD-DN.

Customer Controlled Routing (CCR) "Route to" Command

The Route to destination for 911 calls are restricted to ACD DNs only. If the routing destination is not an ACD DN, the call will be routed to the CDN's default destination ACD DN. CCR calls can be routed to any dialable number.

Enhanced ACD Routing/Customer Controlled Routing

The Enhanced ACD Routing/Customer Controlled Routing (EAR/CCR) features introduce CDNs. The Enhanced ACD Routing (EAR) package 214 allows CDNs to be configured and is a prerequisite of the Meridian 911 (M911) package 224.

INIT ACD Queue Call Restore

INIT ACD Queue Call Restore restores M911 Abandoned calls waiting in either ACD or CDN queues. M911 Automatic Number Identification information is restored on the set display.

Interflow

911 calls interflow the same as other ACD calls. If the interflow feature is configured so that when a call gets busy tone from an internal destination, the 911 call will not get busy tone, but will instead be linked back into the source ACD queue.

If the interflow destination is a number outside the system, the software has no control over the treatment the call gets, so this configuration is not recommended for 911 sites.

Load Management Commands

No changes are made to Load Management for Meridian 911.

Night Service
Night Call Forward

It is recommended that the primary ACD DN not be put in Night Service. If the primary ACD DN is put in Night Service, calls will be sent to the Night Call Forward (NCFW) destination. Even if a 911 call arrived on a trunk with Called Party Disconnect Control (CPDC) defined, the call will still be allowed to NCFW, unlike non-911 ACD calls. This restriction is lifted for 911 calls only. The CWNT set will not ring for calls entering the queue while in Night Service when the queue has a NCFW destination specified.

Overflow

911 calls will overflow (by count and by time) just like any other ACD calls.

Supervisor Control of Queue Size

This feature causes calls to get busy tone once the overflow threshold (OVTH) of the ACD queue is exceeded. This feature is bypassed for 911 calls.

Call Detail Recording (CDR) Records

ANI available for 911 calls is included as the Calling Line Identification (CLID) in CDR Records pertaining to 911-trunk calls. Call Detail Recording records affected are: Normal Records, Start/End Records, Authorization Code Records, Connection Records (Q, R, and F records), and Charge Account Records.

Call Transfer

Trunk priority associated with an incoming 911 call is only preserved if blind transfer is used.

Called Party Disconnect Control

The Called Party Disconnect Control (CPDC) feature is used to retain a 911 trunk when a 911 call is disconnected by the caller. No modification to the feature is required for Meridian 911, except lifting the CPDC and ACD NCFW limitation. 911 calls, arriving through trunks with CPDC defined, will be allowed to NCFW, unlike non-911 ACD calls.

Calling Party Name Display

The Calling Party Name Display feature can be used to configure and display the incoming 911 route name.

Calling Party Privacy

If an incoming call with a Privacy Indicator terminates on a system switch configured with M911, the ANI information (if it exists) is still sent to the Meridian 911 application.

Conference

When a call is answered, and then conferenced, the trunk priority is lost (the conference consultation call is an internal call and treated as low priority by the software). This operation is the same for normal calls and 911 calls.

Dialed Number Identification Service

Dialed Number Identification Service is not supported on 911 trunks.

Display of Calling Party Denied

An incoming M911 call with Automatic Number Identification (ANI) information always displays ANI digits on the terminating set regardless of the calling party's DPD Class of Service.

Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI)

Answering positions are not supported on BRI sets.

Integrated Services Digital Network (ISDN) Primary Rate Interface

911 trunks are not supported on ISDN PRI Trunks or Integrated Service Link (ISL) trunks.

Japan Direct Inward Dialing (DID) Trunks

Japan DID trunks are not supported.

Malicious Call Trace

The Malicious Call Trace (MCT) feature is modified to be supported on ACD sets. ACD sets are allowed to have the Malicious Call Trace Allowed (MCTA) Class of Service and a Trace (TRC) key defined. The feature is activated by pressing the MCT key or dialing an MCT access code.

Malicious Call Trace - Enhanced

The Trunk Hook Flash functionality is used by Meridian 911, Enhanced Malicious Call Trace, and Autodial Tandem Transfer.

No Hold Conference

No Hold Conference calls are treated as internal calls and are linked to the low priority queue of the ACD DN.

Single and Multiple Call Ringing for MADNs

The DN keys for multiple appearance sets can be defined as an SCR (single call ringing) key or as an MCR (multiple call ringing) key. For those DNs (keys on MADN sets) that are SCR, only one call may be answered at a time. That is to say that once a call taker has answered a call, future calls to that DN will receive busy tone until the call taker on that DN has disconnected.

For DNs that are MCR, calls will only be given busy tone once every call taker is busy answering a call. If one call taker is answering a call and there are other call takers available, a new call to that DN will cause the sets of the available call takers to ring. Any available call taker can then answer the new call.

Transfer

Trunk priority associated with an incoming 911 call is only preserved if blind transfer is used.

Meridian 911 Call Abandon

Attendant Break-In

Since an abandoned call does not have a speech path established, the Break-In deny treatment is given to the attendant so that Break-In cannot occur.

Automatic Call Distribution

When a call is abandoned, the call remains in its current state (for instance, Automatic Call Distribution (ACD) queue, CDN queue, or ringing on an ACD agent set).

Automatic Call Distribution Reports

ACD-C and ACD-D packages are not modified for M911 Call Abandon. However, a new interpretation for the report fields are needed for abandoned calls. The incoming call is pegged as an abandoned call when the caller abandons. However, it is not repeatedly pegged as an answered call when the call taker answers the abandoned call.

For ACD-C package, the CALLS ANSWD field only accounts for real calls; the ABANDONED field accounts for abandoned calls that are answered, assuming all abandoned calls are eventually answered by an agent. Consequently, the CALLS ACCPTD field is equal to the CALLS ANSWD field plus the ABANDONED field (number of calls entering queue = number of real calls + number of abandoned ones). This way the Average or Total Call Processing (DCP) Time accurately reflects the amount of time an agent spent on real calls, since answering an abandoned call requires little time. The work an agent does for an abandoned call is more accurately reflected in the DN OUT and OUT TIME fields, which mean total number of outgoing calls and total time of all outgoing calls respectively. Since the agent must hang up the abandoned call and call back to see what the condition is, the outgoing call that is made is more valuable for reporting the agent's work.

For the ACD-D package, the reports also need to be interpreted in this way. When the caller abandons, a CAB message is sent to Meridian MAX; however, later when an abandoned call is answered by an agent no CAA message is sent to Meridian MAX.

Call Force

M911 abandoned calls cannot be call forced.

Call Transfer
M911 abandoned calls cannot be transferred or conferenced.

Called Party Disconnect Control

There is no interaction with M911 Call Abandon and Called Party Disconnect Control.

Conference

M911 abandoned calls cannot be conferenced.

Display Calls Waiting Key
ACD Calls Waiting Key
Ongoing Status Display
Real-time Display

In all of these situations, abandoned calls contribute to the queue count.

Hold

M911 abandoned calls cannot be put on hold.

Initialization

Unanswered abandoned calls are lost if the system initializes.

Interflow

Abandoned calls contribute to the queue count. An abandoned call can interflow only to ACD DNs.

Network ACD

Network ACD is not supported.

Night Service

Abandoned calls can be forwarded to the Night Call Forward DN if the Night Forward DN is an ACD DN. If a primary answering center goes into Night Service while there are abandoned calls in the queue, those abandoned calls are dropped. A CDR N record is printed if CDR is configured.

Night Service Key

Abandoned calls are part of the transition mode when agents go to Night Service and the supervisor selects transition mode.

No Hold Conference

M911 abandoned calls cannot be No Hold conferenced.

Not Ready Key

When an abandoned call is presented to an agent and the agent presses the Not Ready Key, the call is put back into the queue. If an agent is established on an abandoned call and presses the Not Ready Key, the call is dropped.

Overflow by Count

Abandoned calls contribute to the queue count. An abandoned call can overflow.

R2MFC Calling Number Identification/Call Detail Recording Enhancements

M911 trunks do not support Calling Number Identification (CNI). If a CNI is available on an M911 trunk, in addition to the ANI, the ANI is used for the CLID.

Supervisor Observe

Since there is no speech path between the ACD agent and the caller, the supervisor observe feature will be blocked. The supervisor can still press the observe key to observe an agent active on an abandoned call, but will hear silence.

Feature packaging

The following packages are required:

- Digit Display (DDSP) package 19
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package B (ACDB) package 41
- Automatic Call Distribution Package A (ACDA) package 45
- Enhanced Automatic Call Distribution Routing (EAR) package 214
- Meridian 911 (M911) package 224
- Call Waiting Notification (CWNT) package 225

The following additional packages are not required, but are recommended:

- At least one of either Call Detail Recording (CDR) package 4 or Call Detail Recording on Teletype Machine (CTY) package 5
- Automatic Call Distribution Package C (ACDC) package 42 (not needed if packages 51 and 52 are enabled)
- Automatic Call Distribution Load Management Reports (LMAN) package 43
- Automatic Call Distribution Package D (ACDD) package 50
- Automatic Call Distribution Package D, Auxiliary Link Processor (LNK) package 51
- Call Party Name Display (CPND) package 95
- Malicious Call Trace (MCT) package 107
- Calling Line Identification in Call Detail Recording (CCDR) package 118

The M911 Call Abandon feature is included in Meridian 911 (M911) package 224, and requires Call Identification (CALL ID) package 247.

If an application also involves Meridian Link, Meridian Link Module (MLM) package 209 is required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Configure a Terminal Number for an analog (500/2500 type) telephone with a Class of Service of CWNA (Call Waiting Notification Allowed).

2 LD 23 – Configure ACD DNs. The CWNC (CWNT control) is recommended to be set as YES for the primary answering centers (rings for priority calls only) and NO for secondary answering centers (rings for all calls).

3 LD 23 – Configure CDNs. The ceiling value is irrelevant for 911 calls terminating at the CDN, but will be applied to non-911 type calls. When the ceiling value is exceeded, new non-911 calls will receive busy tone.

4 LD 16 – Configure an M911 route.

5 LD 16 – Create a Numbering Plan or Information Digit (NPID) Table:

6 LD 14 – Configure 911 trunks.

7 LD 16 – Configure Call Detail Recording (CDR).

8 LD 17 – Configure the insertion of ANI digits into the CDR record.

9 LD 10 – Configure non-ACD sets (analog (500/2500 type) telephones).

10 LD 11 – Configure non-ACD sets (Meridian 1 proprietary telephones).

11 LD 11 – Configure Meridian 1 proprietary telephones to function as ACD sets.

12 LD 16 – Enable M911 Call Abandon.

13 LD 56 – Configure the new flexible tone for M911 abandoned calls, if desired.

This section provides an example of how to configure Meridian 911. The order in which all items need to be configured to get M911 to run on the system is shown. In addition, the implementation procedures for M911 Call Abandon are shown.

LD 10 – Configure a Terminal Number for an analog (500/2500 type) telephone with a Class of Service of CWNA (Call Waiting Notification Allowed).

Prompt	Response	Description
REQ:	NEW	New.
TYPE:	500	Type of telephone set.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	xxx	Office Data Administration System (ODAS) package designator.
CUST	xx	Customer number, as defined in LD 15
...		
DN	nn...n	Internal Directory Number.
...		
CLS	CWNA	Call Waiting Notification Allowed Class of Service (DTN or DIP).

LD 23 – Configure ACD DNs. The CWNC (CWNT control) is recommended to be set as YES for the primary answering centers (rings for priority calls only) and NO for secondary answering centers (rings for all calls).

Prompt	Response	Description
REQ	NEW	New.
TYPE	ACD	ACD Data Block.
CUST	xx	Customer number, as defined in LD 15
ACDN	nn...n	ACD Directory Number.
...		
MAXP	nn	Maximum number of agent positions.

...		
ISAP	YES	ACD DN uses Meridian Link messaging.
VSID	n	Server ID used for Meridian Link messaging (defined in LD 17).
...		
OVTH	2047	Recommended overflow threshold.
...		
CWNT	l s c u	Call Waiting Notification TN.
CWNC	YES	Call Waiting Notification control.

LD 23 – Configure CDNs. The ceiling value is irrelevant for 911 calls terminating at the CDN, but will be applied to non-911 type calls. When the ceiling value is exceeded, new non-911 calls will receive busy tone.

Prompt	Response	Description
REQ	NEW	New.
TYPE	CDN	Controlled Directory Number Data Block.
CUST	xx	Customer number, as defined in LD 15
CDN	nn...n	Controlled DN number.
...		
DFDN	nn...n	Default ACD DN.
CEIL	2047	Recommended Ceiling Value.
RPRT		Report control.
CNTL	YES	Controlled mode (controlled = YES).
VSID	n	Server ID used for Meridian Link messaging (defined in LD 17).

LD 16 – Configure an M911 route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	DID	Meridian 911 routes use Direct Inward Dialing trunks.
M911_ANI	YES	Enter YES for 911 route.
M911_TRK_TYPE	(911T) 911E	911T = E911 tandem connection. 911E = End office connection.
NPID_TBL_NUM	 0-7	Meridian 911 route table index The ID table must be created before this prompt can be answered.

LD 16 – Create a Numbering Plan or Information Digit (NPID) Table:

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	NPID	Numbering Plan or Information Digit data block.
IDTB	0-7	ID table index. ID table index to be used by this M911 route.
NPID	0-9	NPID for M911 routes.

TRMT	(NONE) NPA FAIL TEST	Numbering Plan Digit or Information Digit treatment.
- NPA	nnn	Numbering Plan Area. Prompted only if TRMT = NPA.

LD 14 – Configure 911 trunks.

Prompt	Response	Description
REQ	NEW	New.
TYPE	DID	Meridian 911 trunks must be DID.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
XTRK	XUT XEM	Universal, or Enhanced E&M trunk card.
CUST	xx	Customer number, as defined in LD 15
NCOS	xx	Network Class of Service Group Number.
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
MNDN	xxxx	Manual Directory Number.
ATDN	xxxxxxx	Autoterminate DN.
TGAR	xx	Trunk Group Access Restriction.
SIGL	EAM EM4 LDR	Trunk signaling.
...		
STRI	WNK	Incoming start arrangement.

SUPN	YES	Answer and disconnect required.
CLS	MFR APY	Meridian 911 trunks must have MFR and APY Classes of Service (this is done automatically).

LD 16 – Configure Call Detail Recording (CDR).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	DID	Meridian 911 routes use DID trunks.
...		
CDR	YES	CDR trunk route.
INC	YES	CDR records generated on incoming calls.
QREC	NO	CDR ACD Q initial records to be generated.

LD 17 – Configure the insertion of ANI digits into the CDR record.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters
...		
- CLID	YES	Calling Line ID (ANI for M911) in CDR.

LD 10 – Configure non-ACD sets (analog (500/2500 type) telephones).

Prompt	Response	Description
REQ:	NEW	Add a set.
TYPE:	500	Type of telephone set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	(DD) SD 4D	(Double), single and quadruple card density.
CUST	xx	Customer number, as defined in LD 15
DIG	xx yy	Dial Intercom Group number and Member number.
DN	nn...n	Directory Number.
...		
IAPG	2	ISDN/AP status message group.
...		
CLS	USMA	Unsolicited Status Allowed Class of Service. M911 position.

LD 11 – Configure non-ACD sets (Meridian 1 proprietary telephones).

Prompt	Response	Description
REQ:	NEW	Add a set.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, and 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	(DD) SD 4D	(Double), single and quadruple card density.

DES	x...x	ODAS set designator.
CUST	xx	Customer number, as defined in LD 15
KLS	1-7	Number of Key/Lamp strips.
...		
CLS	USMA MCTA	Unsolicited Status Allowed Class of Service. M911 position; Malicious Call Trace allowed.
...		
IAPG	2	ISDN/AP status message group.
...		
KEY	xx SCR yyyy	This defines a Single Call Ringing DN key. The xx is the key number and the yyyy is the DN.

LD 11 – Configure Meridian 1 proprietary telephones to function as ACD sets.

Prompt	**Response**	**Description**
REQ:	NEW	Add a set.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, and 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	(DD) SD 4D	(Double), single and quadruple card density.
DES	x...x	ODAS set designator.
CUST	xx	Customer number, as defined in LD 15
KLS	1-7	Number of Key/Lamp strips.
...		

CLS	ADD AGN USMA MCTA	AGN is for agent; SUPN is for supervisor, USMA = M911 position, and MCTA = Malicious Call Trace allowed.
...		
IAPG	2	ISDN/AP status message group.
...		
KEY	0 ACD yyyy	Key 0; ACD; ACD Directory Number.
KEY	xx TRC	Malicious Call Trace key. The xx is the key number.

LD 16 – Enable M911 Call Abandon.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	DID	M911 trunks are DID trunk type.
...		
M911_ANI	(NO) YES	Set to YES to receive ANI for M911 routes.
M911_TRK_TYPE	 (911T) 911E	Meridian 911 ANI trunk types, where: T911T = E911 tandem connections, and 911E = End office connection.

M911_ABAN	(NO) YES	Optional call abandon treatment, where: YES = abandoned call treatment for this route, and NO = no abandoned call treatment for this route.
M911_TONE	(YES) NO	Optional call abandon tone, where: YES = tone given on answer, and NO = silence given on answer.

LD 56 – Configure the new flexible tone for M911 abandoned calls, if desired.

Prompt	Response	Description
REQ	NEW CHG PRT	New, change, or print.
TYPE	FTC	Flexible Tone and Cadence data block.
TABL	0-31	FTC table number.
DFLT	0-31	Default table number.
RING	<CR>	
...		
CAB	YES	M911 Call Abandon upon Answer Tone.
TDSH	i bb cc tt	TDS external, burst, cadence, and tone.
XTON	0-255	NT8D17 TDS Tone code.
XCAD	0-255	NT8D17 cadence code for FCAD.

Feature operation

Meridian 911 operation

To answer a call at a primary, secondary, or alternate answering center that is configured with ACD positions, the 911 call taker presses the ACD DN key. The DN of the incoming call is displayed on the call taker's set.

Meridian 911 Call Abandon operation

When the call is abandoned it remains in its current state (for instance, in CDN or ACD queue or ringing a call taker). Once the call taker answers, a continuous cadenced tone is heard for six seconds, followed by silence. This tone is programmable with the FTC package; otherwise, a default is given. The call taker must hang up and dial the ANI that is shown on the terminal display if call back is required.

Upon answer, the telephone set display is updated with the 911 call taker's ANI and the trunk group name if the Call Party Name Display feature is used. Since the call has been abandoned, the telephone set display flags the abandoned call by appending "ABAND" to the ANI.

Figure 27 shows what is displayed on a telephone set with a Numbering Plan Digit (NPD) call with an NPD of 2 and with the Call Party Name Display feature enabled. The trunk group name is displayed on the first line of the set display; the ANI appears on the second line.

Figure 27
Display for an NPD call

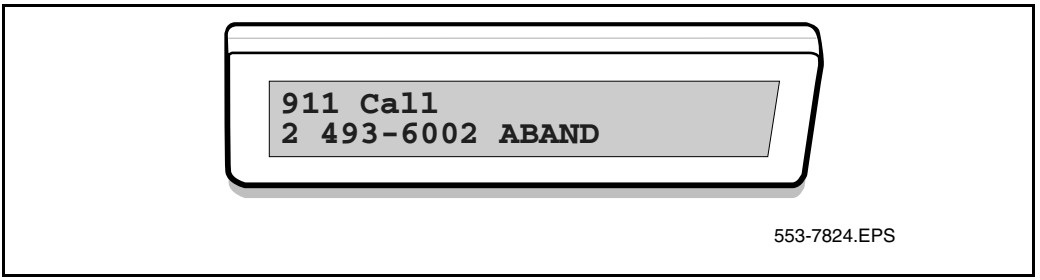
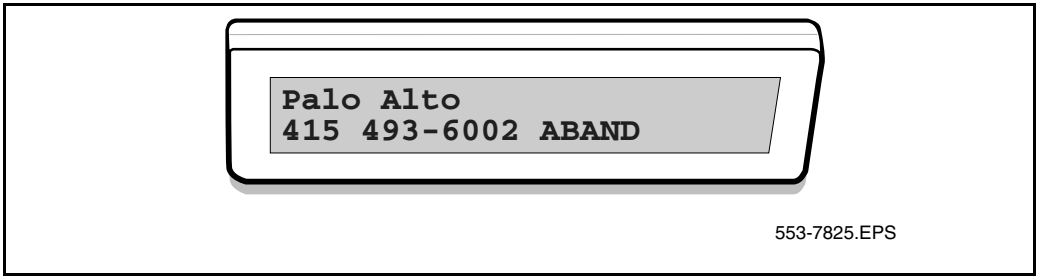


Figure 28 shows a set with an NPA call with an NPD of 1 that was translated to 415 and has the Call Party Name Display feature enabled. The trunk group name (for example, Palo Alto) is displayed on the first line of the set display. The ANI appears on the second line.

Figure 28
Display for an NPA call



Meridian Companion Enhanced Capacity

Contents

This section contains information on the following topics:

Feature description 921
Operating parameters 921
Feature interactions 922
Feature packaging 922
Feature implementation 922
Feature operation 923

Feature description

The Meridian Companion Enhanced Capacity feature doubles the capacity of Meridian Companion and Meridian Companion DECT line cards from 16 units to 32 units. For detailed information, refer to *Meridian Companion* or the *Meridian Companion DECT documentation suite*.

Operating parameters

This feature works with the Companion Meridian Controller Card (CMCC), the Meridian Companion Radio Card (CMRC), the DECT Mobility Card (DMC) and the DECT Mobility Card - Expander (DMC-E). The CMCC and CMRC are supported by all systems with IPE equipment capability. The DMC and DMC-E are supported by all Succession 1000M and Meridian 1 systems.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The Meridian Companion Enhanced Capacity requires the following packages:

- Meridian 1 Companion Option (MCMO) package 240
- MC32 package 350

Feature implementation

This section contains the overlay procedures required to configure the Meridian Companion Enhanced Capacity feature on a system circuit-switched network.

LD 10 – Configure up to 16 or 32 units on a CMCC, CMRC, or DMC/DMC-E.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	500 set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CDEN	(4D)	Card density.
WRLS	(NO) YES	Wireless analog set.
MWUN	(16) 32	Maximum number of wireless units. The MWUN prompt appears only if WRLS = YES.
WTYP	(MCMO) DECT	Wireless type assigns the TN to Meridian Companion cards or to Meridian Companion DECT cards.

Feature operation

No specific operating procedures are required to use this feature.

Meridian Companion / Meridian Companion DECT

Contents

This section contains information on the following topics:

Feature description 925
Operating parameters 926
Feature interactions 926
Feature packaging 926
Feature implementation 927
Feature operation 928

Feature description

Meridian Companion and Meridian Companion DECT are applications on the system that allow digital wireless capabilities. With Meridian Companion or Meridian Companion DECT, users can travel around their work sites while answering a call, making a call, continuing a call, or transferring a call. For detailed information, refer to *Meridian Companion* and the *Meridian Companion DECT documentation suite*.

Operating parameters

Meridian Companion consists of a Companion Meridian Controller Card (CMCC), an optional Meridian Companion Radio Card (CMRC), and an optional Meridian Companion Line card (CMLC). These cards reside on an Intelligent Peripheral Equipment Module of the system. They provide and manage the radio network used in wireless service.

The Meridian Companion DECT includes a DECT Mobility Card (DMC) and a DECT Mobility Card - Expander (DMC-E). These cards exist in an Intelligent Peripheral Equipment Module of the system. The cards provide and manage the radio network used in wireless service.

The Meridian Companion on a Succession 1000M Cabinet or Meridian 1 Option 11C Cabinet, with the Meridian Enhanced Capacity feature and Meridian Companion DR4 (North American) software, can accommodate a maximum of 320 users. Large Systems with Meridian Enhanced Capacity feature and Meridian Companion DR4 (North American) software can accommodate a maximum of 480 users.

The Meridian Companion DECT on a Succession 1000M Cabinet or Meridian 1 Option 11C Cabinet can contain a maximum of 630 users. A Succession 1000M Chassis or Meridian 1 Option 11C Chassis can contain a maximum of 96 users. Large Systems can contain a maximum of 1024 users.

Feature interactions

Meridian Companion supports these features:

- Wireless Privacy, and
- Calling Line Identification (CLID) or Call Party Name Display.

The Meridian Companion DECT does not require DTI programming in LD 73.

Feature packaging

The Meridian Companion and Meridian Companion DECT require the Meridian Companion Option (MCMO) package 240.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Configure a Meridian Companion/Meridian Companion DECT telephone.

2 LD 73 – Configure the Meridian Companion/Meridian Companion DECT pad values.

LD 10 – Configure a Meridian Companion/Meridian Companion DECT telephone.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Analog (500/2500 type) telephone.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
WRLS	(NO) YES	Indicates that this TN corresponds to a portable personal telephone or DECT handset. Only offered if the MCMO package is equipped.
WYTP	(MCMO) DECT	Wireless type assigns the TN to Meridian Companion cards or to Meridian Companion DECT cards. The WYTP prompt appears when WRLS = YES.
CLS	(CNDD) CNDA	Allows the user to see calling or called name associated with the number dialed if CPND is set up for the customer associated with the portable personal telephone. Permitted only if WRLS = YES.

	(MCRD) MCRA	Multiple Call Arrangement (denied) allowed. Allows privacy on analog (500/2500 type) telephones including both portable and wireline sets. Only offered if the MCMO package or SUPP package is equipped.
	(DTN)	Default digit signaling used by portable personal telephone.

LD 73 – Configure the Meridian Companion/Meridian Companion DECT pad values.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	PRI2 PRI	2.0 Mbps/1.5 Mbps PRI data block.
FEAT	PAD	Pad category.
PDCA	1-16	Pad category table.
...		
BRIT	Rx Tx	BRI trunk.
MCM	Rx Tx	Meridian Companion pad value, where: R = Receive T = Transmit, and x = pad value (0-26).

Feature operation

No specific operating procedures are required to use this feature.

Meridian Hospitality Voice Services

Contents

This section contains information on the following topics:

Feature description 929
Operating parameters 930
Feature interactions 931
Feature packaging 932
Feature implementation 934
Feature operation 934

Feature description

Meridian Hospitality Voice Services (MHVS) links Meridian Mail Guest Voice Messaging with the Property Management System (PMS) and the system. Meridian Mail uses information from the Property Management System Interface (PMSI) to manage guest voice messaging and to coordinate the Message Waiting indications for both voice and text messaging.

Meridian Hospitality Voice Services (MHVS) allows Meridian Mail to intercept messages sent over the Property Management System Interface (PMSI) and pass to the system only those messages required to manage and coordinate message indications for both voice and text messages. Should Meridian Mail ever fail, a Meridian Mail bypass switch allows the system to be directly connected to the Property Management System Interface.

Meridian Hospitality Voice Services provides enhancements to the following features:

- Pretranslation
  MHVS will suppress all pretranslation on calls originated by Meridian Mail virtual agents.
- Do Not Disturb
  MHVS allows calls to telephones in a Do Not Disturb (DND) mode to be rerouted to Meridian Mail for special handling.
- Controlled Class of Service (CCOS)
  When CCOS is allowed on M2327 and M3000 telephones, they do not display the softkey choices for standard Meridian Mail features that do not apply when these telephones are used in guest rooms. Dial Access is required to activate these features.

Property Management System (PMS) messages are used to integrate the link.

Operating parameters

The Night Number (NCWF) specified for the AP Recovery enhancement must be local to the system. It cannot be defined using Network Automatic Call Distribution (Network ACD) routing tables.

Attendant Consoles cannot be associated with mailboxes on Meridian Mail.

Softkey menus are suppressed for MHVS commands on M2317 and M3000 telephones when Controlled Class of Service (CCOS) has been activated. Dial Access must be used to operate MHVS features, except guest messaging mailboxes.

When programming the Night Directory Number (Night DN) associated with the customer and Automatic Call Distribution (ACD) queues, be sure to avoid configuring a loopback of Directory Numbers (DNs) for the Night Call Forward DN. For example, if the Night Call Forward DN terminates on a console (directly or indirectly), the attendant Night DN should not terminate on the Meridian Mail virtual ACD DN. With this configuration, calls will remain ringing in the ACD queue under the following conditions:

- The system is in Night Service Mode or
- Meridian Mail fails

The caller remains in the queue until the attendant disengages Night Service, or until the Applications Module Link (AML) recovers from failure.

The use of Integrated Messaging System (IMS) or Integrated Voice Messaging System (IVMS) is not supported with MHVS.

Feature interactions

Attendant End-to-End Signaling

Attendant End-to-End Signaling (AEES), which uses Dual-tone Multifrequency signaling, requires an additional AEES key.

Attendant Overflow Position

Attendant Overflow Position (AOP) allows unanswered calls to the attendant to be forwarded to a customer-defined Directory Number (DN) after a defined time. A call can also be overflowed if all the attendants are in Position Busy State. Overflowed calls can be directed to Meridian Mail. The AOP DN must be defined as an Automatic Call Distribution (ACD) Directory Number (DN), and the ACD DN must have an ACD agent assigned as a virtual VMS agent.

Call Party Name Display

The maximum length of a Call Party Name Display (CPND) name sent from the PMSI/Background Terminal (BGD) is 27 characters. When the full 27-character length is used, part of the CPND name may scroll off the screen. To avoid this problem, the PMSI/Background Terminal (BGD) software has been updated to strip from the screen all trailing blanks from the CPND name.

Centralized Attendant Service

The attendant must be located on the same switch as Meridian Mail for the attendant to use Meridian Mail features.

Digit Key Signaling

Digit Key Signaling (DKS) is supported only from Attendant Consoles at the Meridian Mail site. With DKS equipped, attendants can assist callers in Meridian Mail activities. The attendant can extend source calls to Meridian Mail or direct calls to Meridian Mail.

Do Not Disturb

Individual Do Not Disturb (DND) allows the attendant to place a Directory Number into DND mode. A DN in this mode is free to originate calls, but appears busy to incoming calls. With MHVS equipped, a new prompt (DNDH) allows callers to be redirected to Meridian Mail for voice mail services. A called telephone must have Hunting Allowed (HTA) class of service, and Hunt to Meridian Mail and DNDH in LD 15 must both be set to YES.

M2317, M3000, and Meridian Modular softkey menus

M2317 or M3000 softkey menus are not supported by MHVS. These telephones with Controlled Class of Service Allowed (CCSA) Class of Service are not presented with the Meridian Mail softkey menus when connected to Meridian Mail.

Network ACD

The Night Number (NCFW) specified for the ACD must be local to the node.

Pretranslation

Prior to MHVS, the setup of calls using the Applications Module Link (AML) was not supported from telephones using the Pretranslation feature. With MHVS equipped, call setup using the AML is supported.

Digit Key Signaling
Do Not Disturb Hunt
Message Waiting Indication Interworking
Property Management System Interface

These operations are supported only when Property Management System Interface, Meridian Mail, and attendant and room telephones are located on the same system switch.

Feature packaging

MHVS requires the following packages:

- Meridian Hospitality Voice Services (MHVS) package 179, which requires:

- Recorded Announcement (RAN) package 7
- End-to-End Signaling (EES) package 10
- Make Set Busy (MSB) package 17
- Integrated Messaging System (IMS) package 35
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package A (ACDA) package 45
- Message Center (MWC) package 46
- Command and Status Link (CSL) package 77
- CSL with Alpha Signaling (CSLA) package 85
- Auxiliary Processor Link (APL) package 109

- Property Management System Interface (PMSI) package 103, which requires:
  - Controlled Class of Service (CCOS) package 81
  - Background Terminal Facility (BGD) package 99
  - Room Status (RMS) package 100

Attendant Overflow Position (AOP) package 56 is required for AOP Directory Number (DN) enhancement.

- Digit Key Signaling (DKS) package (180), which requires:
  - Meridian Hospitality Voice Services (HVS) package 179
  - The site may also require other packages, such as:
  - Message Registration (MR) package 101
  - Automatic Wake Up (AWU) package 102

Feature implementation

Refer to *Meridian Mail Modular Option Guest Voice Messaging* (553-7041-210) and *Hospitality Features: Description and Operation* (553-3001-353).

Feature operation

Refer to *Meridian Mail Modular Option Guest Voice Messaging* (553-7041-210) and *Hospitality Features: Description and Operation* (553-3001-353).

Meridian Mail Trunk Access Restriction

Contents

This section contains information on the following topics:

Feature description 935
Operating parameters 938
Feature interactions 938
Feature packaging 939
Feature implementation 939
Feature operation 940

Feature description

The Meridian Mail Trunk Access Restriction (MTAR) feature prevents direct or indirect call transfer or conference of external calls to Meridian Mail. In this feature, external calls are defined as incoming/outgoing trunk calls that originate or terminate outside a private network.

This definition is applicable to all types of trunks, with the exception of TIE trunk calls. External calls are separated from a transferring/conferencing set on a network using TIE trunks. MTAR operation is dependant on the information sent to the remote node from the node that is attempting to transfer/conference. MTAR is triggered if the network information (such as Network Attendant Service or Calling Line Identification) indicates that an external call and a transfer/conference attempt to Meridian Mail is occurring. MTAR is also triggered if local information, such as Route Class, indicates an external call and a transfer/conference attempt to Meridian Mail is occurring.

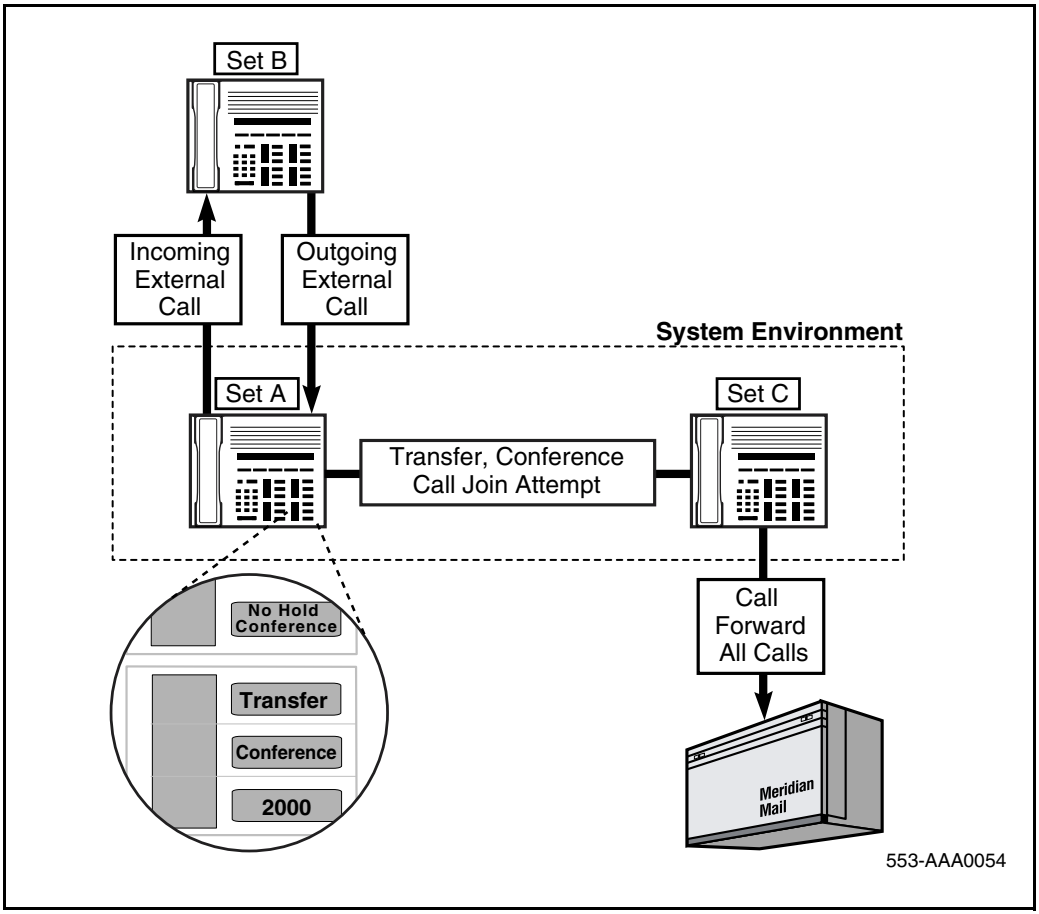
Meridian Mail Trunk Access Restriction averts potential Meridian Mail system abuse by distinguishing between internal and external calls that are directed to Meridian Mail. When activated, Meridian Mail Trunk Access Restriction impacts the operation of the following features:

- Call Transfer
- Conference
- No Hold Conference
- Call Join capabilities of the Multi-Party Operation feature

Meridian Mail Trunk Access Restriction prevents the completion of any Call Transfer, Conference, No Hold Conference or Call Join attempts on incoming/outgoing external calls to Meridian Mail.

As illustrated in Figure 65, MTAR's capabilities prevent an established call between Set A, an internal call, and Set B, an external call, from being forwarded to Meridian Mail. When Set A attempts to either Transfer, Conference, No Hold Conference or Call Join Set B to Set C, which is either a direct Meridian Mail DN or has activated Call Forward All Calls (CFAC) to Meridian Mail, the transfer and conference keys are ignored when pressed to complete operation.

Figure 65
Meridian Mail Trunk Access Restriction Call Transfer



Operating parameters

MTAR does not treat Centralized Attendant Position and Night Attendant sets as Attendant Consoles. These sets receive treatment based on their actual set type. For example, if the night attendant is a proprietary set, then it is treated as a proprietary set.

The operation of an Attendant Console is not affected when this feature is enabled. An attendant can transfer or conference an external line to Meridian Mail directly or indirectly.

MTAR does not affect Automatic Attendant, Customer Controller Routing, Integrated Voice Response or Meridian Link features. However, if a user disallows any of these features from accessing Meridian Mail, the application must be written to take this into account.

Call transfer from ISDN Basic Rate Interface (BRI) set is not supported.

In a networking environment, Meridian Mail must reside on the same node as the transferring/conferencing set.

Feature interactions

Traffic Reporting

Traffic Reporting and Optivity Telephony Manager's (OTM) traffic report, TFC005, are modified to report the number of times this feature is requested. A new line is added for the Meridian Mail Trunk Access Restriction which is identified by the feature number "27" and its peg count.

Network Call Transfer
Network Call Conference

Meridian Mail Trunk Access Restriction (MTAR) requires the transferring or conferencing set and Meridian Mail to be located on the same node. If the transferring or conferencing set are located not on the same node as Meridian Mail, the MTAR feature is not provoked because the call transfer/ conference attempt is terminated by a network on Meridian Mail.

However, an external call can be transferred or conferenced over the network, using TIE trunks. This operation is dependant on the type of network information the remote node forwards to the node where the transfer/ conference attempt is made. Meridian Mail must be on the transferring/ conferencing node. If network information is provided, indicating that an external call is attempting to transfer/ conference to Meridian Mail, the MTAR feature is invoked. When no network information is provided, MTAR is provoked if the local information (Route Class) indicates that an external call to Meridian Mail is being attempted.

Feature packaging

Meridian Mail Trunk Access Restriction requires Message Waiting Center (MWC) package 46.

Feature implementation

Meridian Mail Trunk Access Restriction feature requires prior installation of Meridian Mail. The implementation of this feature, therefore, assumes that Meridian Mail has been properly configured.

LD 15 – Enable Meridian Mail Trunk Access Restriction.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Customer Features and Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	MCI	Message Centre Included.
...		
- MTAR	YES	Meridian Mail Trunk Access restricted. NO = Meridian Mail Trunk Access allowed.

Feature operation

Call Transfer/Conference

Proprietary Set

Set A is a proprietary set with Transfer Key and Conference Key.

1 An incoming/outgoing external call is established between Set A and Set B, an external party. The call between Set A and Set B is active on Key X.

2 Set A presses the Transfer/Conference Key that automatically puts Set B on Hold.

3 Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail or is a Meridian Mail DN.

4 When Set A attempts to transfer/conference Set B by pressing more than once the Transfer/Conference Key it is ignored.

5 Set A recovers Set B by pressing Key X.

No Hold Conference
Proprietary Set/ISDN BRI Set

When Meridian Mail Trunk Access Restriction is enabled, direct or indirect no hold conference to an external call is permitted. During direct or indirect no hold conference, the calling party is never put on hold.

Set A is a proprietary set or an ISDN BRI set with a No Hold Conference Key configured as either No Hold Conference, Conference Autodial, Conference Speed or Conference Hotline.

1 An incoming/outgoing external call is established between Set A and Set B, an external party. The call between Set A and Set B is active on Key X.

2 Set A presses the No Hold Conference Key.

3 Set A dials Set C. Set C is has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.

4 The conference is set up as normal. However a two party connection between Set B, an external party, and Meridian Mail is not allowed if the call controller releases. If this occurs, the connection between the trunk and Meridian Mail party is dropped.

Transfer
Analog (500/2500 type) Set

Set A is an Analog (500/2500 type) set with a XFA Class of Service (transfer and three/six party conference allowed).

1 An incoming or outgoing external call is established between Set A and Set B, an external party.

2 Set A performs a switchhook flash that puts Set B on hold.

3 Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.

4 Before or after the Meridian Mail has answered, Set A attempts to transfer Set B to Meridian Mail by going on-hook.

5 This attempt is treated as an illegal transfer. Set A is re-rung and reconnected with Set B when going off-hook.

Conference
Analog (500/2500 type) Set

Set A is an Analog (500/2500 type) set with a XFA Class of Service (transfer and three/six party conference allowed).

1 An incoming or outgoing external call is established between Set A and Set B, an external party.

2 Set A performs a switch hook flash that puts Set B on hold.

3 Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.

4 Before or after the Meridian Mail has answered, Set A attempts to conference Set B to Meridian Mail by performing another switchhook flash.

5 The conference is not permitted. Set A is reconnected to Set B. The call to Meridian Mail is disconnected.

Set A is an Analog (500/2500 type) set with a TSA Class of Service (three party service allowed).

1 An incoming or outgoing external call is established between Set A and Set B, an external party.

2 Set A perform a switch hook flash that puts Set B on hold.

3 Set A dials Set C. Set C has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.

4 Before or after Meridian Mail has answered, Set A attempts to conference Set B to Meridian Mail by dialing the conference control digits.

5 The conference is not permitted and Set A is reconnected to Set B. The call to Meridian Mail is disconnected.

Table 80 summarizes how different external calls are handled when Meridian Mail Trunk Access Restriction is enabled.

Table 80
Summary of Meridian Mail Trunk Access Restrictions

Telephone	External Call Type	Operation	Failure Treatment	Result
500/2500	Incoming	Transfer to Meridian Mail (MMail)	Re-ring to transferring set	Not allowed
500/2500	Incoming	Transfer to set with Call Forward All Calls (CFAC) to MMail	Re-ring to transferring set	Not allowed
500/2500	Outgoing	Transfer to MMail	Disconnect external call and Meridian Mail	Not allowed
500/2500	Outgoing	Conference to set with CFAC to MMail	Disconnect external call and Meridian Mail	Not allowed
500/2500	Outgoing/ Incoming	Conference to MMail	Reconnect to external call. Disconnect call to MMail	Not allowed

500/2500	Outgoing/ Incoming	Conference to set with CFAC to MMail	Reconnect to external call. Disconnect call to MMail	Not allowed
Proprietary	Outgoing/ Incoming	Transfer/ Conference to MMail	Operation ignored	Not allowed
Proprietary	Outgoing/ Incoming	Transfer/ Conference to set with CFAC to MMail	Operation ignored	Not allowed
Proprietary	Outgoing/ Incoming	No Hold Conference to Meridian Mail or to set CFAC to MMail	Not applicable	Allow
Proprietary	Outgoing/ Incoming	No Hold Conference release to make MMail to trunk two-party connection	Disconnect MMail and external trunk	Not allowed
Proprietary	Outgoing/ Incoming	Call Join of external call to MMail	Operation ignored	Not allowed
Basic Rate Interface	Outgoing/ Incoming	Conference to MMail	Operation ignored	Not allowed
Basic Rate Interface	Outgoing/ Incoming	Conference to set with CFAC to MMail	Operation ignored	Not allowed
Attendant	Outgoing/ Incoming	Transfer/ Conference to Meridian Mail	Not Applicable	Allowed
Attendant	Outgoing/ Incoming	Transfer/ Conference to set with Call Forward All Calls to Meridian Mail	Not Applicable	Allowed

Meridian Mail Voice Mailbox Administration

Contents

This section contains information on the following topics:

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Feature description

The Meridian Mail Voice Mailbox Administration (VMBA) feature enables the system administrator to use system administration overlays to administer and maintain the Meridian Mail Voice Mailbox Application. This feature streamlines the process of implementing and maintaining voice mailboxes (VMBs).

VMBA provides the following capabilities:

- Accessing the Voice Mailbox Application using LDs 10 and 11 rather than through a separate terminal

- Viewing application and mailbox statistics to help ensure the integrity of the application
- Synchronizing the system and Meridian Mail databases using special audit and upload functions:
  - The audit function helps ensure that name data stored on the system is synchronized with name data stored on Meridian Mail. The system administrator can run the audit manually or request that the system run it periodically.
  - For sites that want to implement VMBA and already have VMBs configured on Meridian Mail, the VMBA upload function lets the system administrator create or update the system VMB database from the existing Meridian Mail VMB database. Upload can significantly reduce the time required to implement VMBA.

Access to Meridian Mail VMB administration functions is still available with the Meridian Mail administration console. However, to prevent database inconsistencies, use the system for VMB administration when VMBA is equipped.

CAUTION

There is a potential impact on the CPND database when using the VMBA application. Therefore, users should read with care the sections entitled "Name processing considerations" on page 949 and "Site with a preconfigured Meridian Mail database" on page 958.

Operating parameters

The appropriate VMB Class of Service must be defined on Meridian Mail before the system can add VMBs. Otherwise, Meridian Mail transaction errors will occur. A Meridian Mail Class of Service specifies a particular set of Meridian Mail options.

A system supports only one Meridian Mail system for VMBs.

The system allows for only one VAS and one customer to be configured for this application.

If a VMB is deleted on the system but not on Meridian Mail, the result could be an orphan VMB. If the DN for the deleted VMB is reused on the system, Meridian Mail deletes the old DN and adds the new one, thereby recovering the associated VMB. If the DN is not reused, the orphan VMB is not recovered.

VMB changes made directly on a Meridian Mail administration terminal may not be detected for up to five days, because system automatic database audits (if equipped) can only run every five days.

The VMB status printed in LD 20 indicates the status of transactions on the system, not on Meridian Mail. For example, if a VMB is disabled on Meridian Mail, its state is not updated on the system.

VMBs cannot be configured for telephones served by a remote Meridian Mail subsystem.

A VMB is not affected when a user's telephone is disabled or being relocated. The VMB remains logged in and continues to receive incoming messages.

Feature interactions

Automatic Set Relocation

Relocating a user with an associated VMB to a new TN will not affect the VMB. The VMB remains logged in and continues to receive incoming voice messages while the telephone is being relocated.

A telephone that is relocated out but not relocated back in can still have an active VMB. A relocated telephone must be deleted manually on the system before its associated VMB is removed.

Call Waiting Redirection

Unanswered calls given Call Waiting treatment may now be allowed to forward to Voice Mail through the activation of the Call Waiting Redirection feature. The greeting given to the caller is for a "no answer" condition.

Call Party Name Display

There is significant interaction between the Call Party Name Display (CPND) database and the Meridian Mail VMB database. The sections entitled "Common data elements" on page 948 and "Name processing considerations" on page 949 describe these interactions.

Meridian Mail

Although there is no user impact, unsolicited link messages will appear when VMBA is equipped.

Common data elements

Table 81 shows the data that is stored and synchronized between the system and Meridian Mail.

Table 81
Data stored by both the system and Meridian Mail

System	Meridian Mail	Description
DN	Mailbox number	System DN to which a VMB is assigned
VMB Class of Service	Class of Service	Specific set of Meridian Mail options
CPND name	First name/Last name/ Initial	Name associated with a VMB (optional)
Second DN	Second DN	Second DN sharing a mailbox (optional)
Third DN	Third DN	Third DN sharing a mailbox (optional)

VMB data configured on the system and downloaded to Meridian Mail is subject to the same validation routines as data entered directly at the Meridian Mail administration terminal. When downloaded VMB data fails Meridian Mail validation, a message prints on the system TTY.

Name processing considerations

There are basic differences in how the CPND and Meridian Mail process name data. This section describes those differences and makes specific recommendations for minimizing their impact on your system.

Because this feature may affect your name data, print the system and Meridian Mail name databases before beginning to implement VMBA on a system with VMBs already implemented. (Use the appropriate administrative overlays to print the databases.)

Name lengths

System versus Meridian Mail

Because the allowable name lengths differ between Meridian Mail and the system, it is recommended that you use the most restrictive case for name lengths on both systems.

Meridian Mail accepts the following name lengths:

- Up to 21 characters for first name
- Up to 40 characters for last name, and
- Up to 61 characters for combined first and last names.

CPND accepts the following name lengths:

- Up to 27 characters for first name
- Up to 27 characters for last name, and
- Up to 27 characters for combined first and last names.

When the VMBA application is installed, the recommended name lengths on both the system and Meridian Mail are as follows:

- Up to 21 characters for first name. Meridian Mail truncates a system first name that is longer than 21 characters.

- Up to 27 characters for combined first and last names. If names on Meridian Mail exceed a combined length of 27 characters, they are truncated on the system during an upload.
- Up to 27 characters for last name. Last names are truncated to 27 characters when uploaded.

Name handling during an upload

If the CPND package is equipped and CPND is configured for the customer, the following name processing occurs during an upload:

1 If a name already exists on the system, it is replaced with the uploaded name using the expected length (XPLN) and display formats configured for that name.

2 If a name does not exist on the system, the uploaded name is added using the default length (DFLN) specified for the customer and the default display format of FIRST, LAST.

3 If the names received from Meridian Mail are longer than the expected or default length, the first name is truncated until both names fit into the configured length. If necessary, the last name is also truncated.

For example, if Meridian Mail sends the name JACK FROST and XPLN is 8, the name is truncated to JA FROST. If XPLN is 4, the name is truncated to FROS.

A subsequent audit with DATA_CORRECT set to ON causes the name on Meridian Mail to be updated with the system name (either JA FROST or FROS).

Character sets

Meridian Mail supports a subset of the characters that the system supports. When Meridian Mail encounters a name from the system that contains characters outside its supported character set, it rejects the name. Therefore, it is recommended that you use the most restrictive character set.

The character sets supported by the system and Meridian Mail are as follows:

- System: ASCII H.20 through H.7E, excluding asterisk (*) and exclamation point (!)

- Meridian Mail: ASCII H.20 through H.7E excluding the plus sign (+), underscore (_), and question mark (?)

Therefore, on a system with VMBs, the system user should avoid using the asterisk (*), exclamation point (!), plus sign (+), underscore (_), and question mark (?) in CPND names.

Database synchronization considerations

As you configure and implement VMBA, keep the following points in mind:

- System and Meridian Mail each has its own name database. Therefore, to ensure synchronization, enter and change name information from the system only. VMBA facilities ensure that corresponding changes are made to the Meridian Mail database. However, remember that changes made directly to the Meridian Mail are not made to the system database.
- The VMBA Audit facility not only detects VMB database mismatches; with Data Correction enabled, the Audit facility invokes processing to make the Meridian Mail VMB database match the system VMB database. See Table 82.

Table 82
Effect of running Audit with Data Correction enabled

Status of VMB		Effect on VMB databases	
System	**Meridian Mail**	**System**	**Meridian Mail**
VMB not configured	VMB not configured	No change	No change
VMB not configured	VMB configured	No change	No change
VMB configured	VMB not configured	No change	VMB added
VMB configured	VMB configured; database matches system	No change	No change
VMB configured	VMB configured; database does not match system	No change	VMB database changed to match system database

- The VMBA Upload facility forces the system VMB database to match the Meridian Mail VMB database. In the case where VMB is not configured on Meridian Mail, an upload will delete the system VMB database. See Table 83.

Table 83
Effect of running Upload

Status of VMB		Effect on VMB databases	
System	**Meridian Mail**	**System**	**Meridian Mail**
VMB not configured	VMB not configured	No change	No change
VMB not configured	VMB configured	VMB added	No change
VMB configured	VMB not configured	VMB deleted	No change
VMB configured	VMB configured; database matches system	No change	No change
VMB configured	VMB configured; database does not match system	VMB database changed to match Meridian Mail database	No change

Feature packaging

Meridian Mail Voice Mailbox Administration (VMBA) is available as package 246.

Although not required, Calling Party Name Display (CPND) package 95 for the system is recommended. Certain Meridian Mail features, such as name dialing, require that CPND be equipped.

Alarm Filtering package 243 is recommended because of the additional information that appears in the formatted output.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 17 – Configuring the VMBA application.
2. LD 10 – Add a VMB on an analog (500/2500 type) telephone.
3. LD 11 – Add a VMB on a Meridian 1 proprietary telephone.
4. LD 20 – Print the DN block.
5. LD 20 – Print the TN block.
6. LD 20 – Print VMB data.
7. LD 83 – Print ODAS data.

Be sure to print the name databases for both the system and Meridian Mail before beginning to implement the VMBA application.

Implementing VMBA requires that it be installed and equipped on the system. (In addition, Meridian Mail must be MM9 or later.) This section includes instructions for three implementation scenarios:

1. A site with no preconfigured database on either the system or Meridian Mail.
2. A site with a preconfigured database on the system, but not on Meridian Mail.
3. A site with VMBs configured on Meridian Mail, but not on the system.

Site with no preconfigured database

1. If necessary, configure and enable the AML link to Meridian Mail.
2. Configure the VMBA application in LD 17 on the VAS link associated with Meridian Mail. Set the DATA_CORRECT and AUTO_AUDIT options to ON to simplify database maintenance and ensure data integrity.

LD 17 – Configuring the VMBA application.

Prompt	Response	Description
REQ	NEW	Add.
TYPE	VAS	Value Added Server
VAS	NEW CHG	Add or change a value added server link.
- VSID	0-15	VAS identifier.
- AML	0-15	Application Module Link identifier.
- APPL	NEW VMBA	Configure the VMBA appli.cation associated with a VSID.
- CUST	xx	Customer number, as defined in LD 15
- - DATA_CORRECT	ON	Enable automatic database correction during audit; the Meridian Mail database is updated to match the system database.
- - AUTO_AUDIT	ON	Enable automatic database audit; the Meridian Mail database is audited every 5 days as part of daily routines.

If the AML link is active, the VMBA application is automatically enabled after it is configured in LD 17. If the AML link is not active, the VMBA application is placed in the LINKOOS (link out of service) status.

3 Configure the VMB Classes of Service on Meridian Mail. Transaction errors occur if a Class of Service specified on the system has not been configured on Meridian Mail.

4 Use LDs 10 and 11 to administer VMBs on the system. The database changes are automatically downloaded to Meridian Mail if both the AML and the VMBA application are enabled. If either is disabled, the VMBs that are added or changed are left in the UPDATE PENDING state. They are downloaded when both the AML link and the application are enabled.

LD 10 – Add a VMB on an analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	500 2500	DN related data.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
DN	xxxx	Directory number.
- MARP	YES	Multiple Appearance Redirection Prime.
- CPND	NEW CHG	Gateway to change Calling Party Name Display data.
- - VMB	NEW CHG	Gateway to change VMB data associated with the above DN.
- - VMB _COS	0-127	VMB class of service; must already be defined on Meridian Mail to avoid transaction errors.
- - SECOND _DN	xxx...x	Second DN sharing this VMB. To delete a DN, enter X <CR>.
- - THIRD _DN	xxx...x	Third DN sharing this VMB. To delete a DN, enter X <CR>.
- - KEEP _MSGS	(NO) YES	For a new VMB only, indicates whether messages and current password on Meridian Mail should be preserved if a VMB with the same DN already exists.

LD 11 – Add a VMB on a Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	NEW	Add.
TYPE:	aaaa	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
KEY	xx yyy zzzz	Telephone function key assignments.
- MARP	YES	Multiple Appearance Redirection Prime.
- CPND	NEW CHG	Gateway to Calling Party Name Display data.
- - VMB	NEW CHG	Gateway to change VMB data associated with the above DN.
- - SECOND _DN	xxx...x	Second DN sharing this VMB. To delete a DN, enter X <CR>.
- - THIRD _DN	xxx...x	Second DN sharing this VMB. To delete a DN, enter X <CR>.
- - KEEP _MSGS	YES (NO)	For a new VMB only, indicates whether messages and current password on Meridian Mail should be preserved if a VMB with the same DN already exists.

Site with a preconfigured system database

Typically, this scenario involves a new system installation. The database is created on the system and subsequently downloaded when the AML link and Meridian Mail are operational.

Configuring the database

1. Configure the VMBA application in LD 17 on the VAS associated with Meridian Mail. Set the DATA_CORRECT and AUTO_AUDIT options to OFF until the installation is complete.

   The AML link does not have to be configured at this point because there is no actual hardware to enable.

2. Configure the telephones and associated VMBs. The VMBs will be left in UPDATE PENDING state.

Installing the database at the customer site

1. Ensure that the Meridian Mail database is configured with the VMB Classes of Service that were used when configuring the system database. *Do not proceed with step 2 until this step is completed.*

2. If necessary, configure and enable the AML link to Meridian Mail.

3. Unless the VMBA application is in a manually disabled state, it will be automatically enabled. If it is manually disabled, use LD 48 to enable it. See “Enabling the VMBA application” on page 959.

4. When the VMBA application is enabled, the system will begin downloading the preconfigured database to Meridian Mail. Use the PRT VMB option in LD 20 to monitor the progress of the download.

5. After the download is complete, check the system TTY for errors and make corrections manually.

6. Use LD 48 to initiate a manual audit of the entire database. This is to verify that the VMB and CPND data on the system matches the downloaded data on Meridian Mail. See “Starting a manual audit” on page 968.

   To determine the status of the audit, use the STAT VMBA <vsid> AUDT command in LD 48. When the audit is complete, check the audit report for errors; make corrections manually.

7. Configure the DATA_CORRECT and AUTO_AUDIT options as desired. It is recommended you set them to ON to help ensure database integrity.

   Installation is now complete. Use the system to perform ongoing administration of VMBs.

Site with a preconfigured Meridian Mail database

Existing sites installing the VMBA application may have VMBs already configured on Meridian Mail. LD 48 includes an upload option that simplifies VMB data configuration on the system.

CAUTION

The upload option also causes name data configured on Meridian Mail to be uploaded to the system. Any existing names on the system are replaced with names currently configured on Meridian Mail. See "Name processing considerations" on page 949 for an explanation of the changes that may result.

1. If necessary, configure and enable the AML link to Meridian Mail.
2. Configure the VMBA application in LD 17 on the VAS associated with Meridian Mail. Set the DATA_CORRECT and AUTO_AUDIT options to OFF until the installation is complete.

   If the AML link is active, the VMBA application is automatically enabled after it is configured in LD 17. If the AML link is not active, the VMBA application is placed in the LINKOOS (link out of service) state.
3. Initiate the database upload by entering the following command in LD 48:

   ENL VMBA <vsid> UPLD ALL

   To check the status of the upload, enter the following command in LD 48:

   STAT VMBA <vsid> UPLD
4. When the VMB UPLOAD COMPLETE message appears, investigate and resolve any errors that occurred during the upload.
5. Initiate a manual database audit using the following command in LD 48:

   ENL VMBA <vsid> AUDT ALL

   This will verify that the VMB and CPND data on the system matches the data on Meridian Mail.

6 Manually resolve any errors detected by the audit. Perform any necessary name cleanup.

7 Configure the DATA_CORRECT and AUTO_AUDIT options as desired. It is recommended you set them to ON to help ensure database integrity.

Installation is now complete. Use the system to perform ongoing administration of VMBs.

Feature operation

Enabling the VMBA application

Use the VAS gateway in LD 17 to configure the VMBA application. After configuring the VMBA application, the system sets the VMBA application state to INACTIVE and immediately attempts to establish a VMBA session with Meridian Mail. If successful, the system changes the VMBA application state to ACTIVE and prints an APPLICATION ENABLED message on the TTY. If unsuccessful, the following actions occur:

- If the AML link is down:
  - The system issues a “FAILED TO ENABLE APPLICATION” message to the TTY.
  - The application’s state is changed to LINKOOS (link out of service).
  - The application is automatically enabled when the link becomes available.
- If the AML link is up but the application is not responding on Meridian Mail:
  - The system attempts to establish a session every two minutes until successful or until the user disables the application using LD 48.
- If the AML link is up but the application is not equipped on Meridian Mail:

— For MM8 and earlier Releases, the system attempts to establish a session as described above. Such attempts fail. Disable VMBA until the upgrade to MM9 occurs.

— For MM9 and later Releases, Meridian Mail indicates to the system that the feature is not configured. The message "FAILED TO ENABLE APPLICATION" appears on the TTY, indicating that the request is rejected. The application remains in INACTIVE status. Retries continue until the user disables the application in LD 48 or until the application is equipped on MM9.

If the VMBA application is not automatically enabled, use the following command in LD 48 to enable it:

ENL VMBA <vsid>

where <vsid> is the VAS identifier, in the range of 0-15.

Disabling the VMBA application

LD 48 accepts the following command to disable the VMBA application:

DIS VMBA <vsid>

where <vsid> is the VAS identifier, in the range of 0-15.

The following actions occur when the application is disabled:

1 The VMBA application state is changed from ACTIVE to MANDIS.

2 All VMB transactions in progress with Meridian Mail are aborted. VMBs defined on the system but not successfully updated on Meridian Mail remain in the UPDATE PENDING state. They will be processed when the application is reenabled.

3 Database audit or upload activities are aborted.

4 The VMBA session established with Meridian Mail is released.

Determining the status of the VMBA application

LD 48 accepts the following command to print the status of the VMBA application:

STAT VMBA <vsid>

where <vsid> is the VAS identifier, in the range of 0-15.

Output from this command, shown in the following example, indicates the status of the application, the audit function, and the upload function:

VMBA ACTIVE
AUDIT INACTIVE
UPLOAD INACTIVE

Valid application states for VMBA appear in Table 84.

Table 84
VMBA application states

State	Explanation
INACTIVE	The application has been configured in LD 17 but is inactive for one of the following reasons: — An application session request was sent to Meridian Mail but confirmation has not yet been received. — Meridian Mail is not configured to support the VMBA application (it does not have the application equipped, or it is running on MM8 or earlier). — A “FAILED TO ENABLE APPLICATION” message on the TTY indicates a reason why the application is inactive.
MANDIS	The application was manually disabled using LD 48.
LINKOOS	The application is inactive because the link to Meridian Mail is out of service.
ACTIVE	The application is enabled and operational.

Managing voice mailbox data

Adding or changing a VMB

Use LDs 10 and 11 to add or change a VMB. Use LD 10, 11, or 95 to add or change a name.

When a VMB is added or changed, the system places the VMB in the UPDPEND (update pending) state and informs a background process that an update is pending. The background process initiates an update transaction with Meridian Mail, with one of these outcomes:

- The operation is successful; the VMB state becomes CONFIGURED.
- The operation fails (perhaps because of bad data); the VMB state becomes UPDFAIL (update failed) and a technician must manually intervene to correct the error condition.
- If the VMB already exists on Meridian Mail when the system requests a VMB add, one of the following outcomes results:
  - — If the response to the KEEP_MSGS prompt in LDs 10 and 11 was NO, Meridian Mail deletes the existing VMB and creates a new one using the configuration information specified by the system. All existing messages and passwords are deleted.
  - — If the response to the KEEP_MSGS prompt in LDs 10 and 11 was YES, Meridian Mail keeps all existing messages and passwords associated with the VMB, but replaces the existing configuration information with the new configuration specified by the system. This information includes user name, Class of Service, and so forth. Meridian Mail automatically enables newly created VMBs.

Deleting a VMB

There are three ways to delete a VMB:

- When using LDs 10 and 11, enter OUT at the VMB prompt.

  When doing a normal CHG or ECHG on a telephone in LDs 10 and 11, enter OUT at the VMB prompt to delete the telephone's VMB.

- When using LDs 10 and 11 to delete a telephone, enter OUT at the REQ prompt.

  If a telephone is configured with a Single Appearance DN, the DELETE_VMB prompt appears after the technician enters OUT at the REQ prompt. A YES response causes the VMB to be deleted on both the system and Meridian Mail. A NO response causes the VMB to be deleted on system but not on Meridian Mail.

  The DELETE_VMB and the KEEP_MSGS prompts allow a technician to move a user from one telephone type to another without having to delete and re-create the VMB.

  — DELETE_VMB = NO when deleting a DN keeps the old mailbox. KEEP_MSGS = YES when adding a new telephone (with the old, previously deleted DN) keeps the VMB messages and password from the old DN intact.

  — DELETE_VMB = NO when deleting a DN keeps the old mailbox. KEEP_MSGS = NO when adding a new telephone (with the old, previously deleted DN) deletes the VMB messages and password associated with the mailbox.

- When changing a Single Appearance DN on a telephone, the system automatically deletes the old DN and associated VMB.

  When the changed DN is entered, if it is currently assigned to another telephone that has a VMB associated with it, the telephone with the changed DN becomes a user of that VMB. If the changed DN does not currently have a VMB, one can be added.

  Note: When changing the DN for a member of a Multiple Appearance DN group, the VMB for the Multiple Appearance DN is unaffected.

Printing VMB data

LDs 20 and 83 support printing VMB data associated with a telephone. LDs 10 and 11 can access LD 20 to facilitate printing VMB data after it is entered.

LD 20 provides three ways to print VMB data:

- Use the PRT DNB command to print the DN block.

LD 20 – Print the DN block

Prompt	Response	Description
REQ	PRT	Print.
TYPE	DNB	DN related information.
CUST	xx	Customer number, as defined in LD 15
DN	xxxx	Directory Number.

- Use the PRT TNB command to print the TN block.

LD 20 – Print the TN block

Prompt	Response	Description
REQ	PRT	Print.
TYPE	TNB aaaa	TN block, or any telephone configured in LD 11.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

- Use the PRT VMB command to print the VMB DN and VMB state. For a definition of each state, see Table 85.

LD 20 – Print VMB data.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	VMB	VMB related information.
CUST	xx	Customer number, as defined in LD 15
DN	xxxx xxxx-yyyy (ALL)	Print data for a single DN. Print data for a range of DNs. Print data for all DNs with VMBs.
VMB_STATE	(ALL) UPDPEND CONFIGURED UPDFAIL MISMATCH UPDINPROG INVALID	Print all VMBs regardless of state. Print VMBs in update pending state. Print configured VMBs. Print VMBs whose updates failed. Print VMBs with database mismatches. Print VMBs with updates in progress. Print VMBs in an invalid state.

Table 85
VMB states

State	Explanation
CONFIGURED	The VMB is configured on the system and Meridian Mail.
UPDPEND	A VMB update is pending. The VMB has been added or changed on the system but Meridian Mail has not yet been updated. When the AML link comes up (if it is down), or when the backlog of updates (if any) is processed, the VMB will be updated automatically.
UPDINPROG	A VMB update is in progress. The request was sent to Meridian Mail but a confirmation has not yet been received by the system.
UPDFAIL	A transaction with Meridian Mail failed. A VMB UPDATE FAIL error message appears on the system TTY indicating the cause of the failure. A technician must intervene to correct the problem.
MISMATCH	There is a database mismatch between the system and Meridian Mail. The mismatch was detected by VMBA Audit but not corrected (because database correction is not enabled in LD 17). A “VMB MISMATCH FOUND” error appears on the system TTY indicating the mismatch. A technician must intervene to correct the problem.
INVALID	The VMB is in an invalid state. Verify that the VMB data for the DN is correct on the system. Then use LD 48 to run VMB Audit on the DN.

To print VMB data in LD 83, respond with TNB at the REQ prompt. This response causes the TN block to print, including VMB data.

LD 83 – Print ODAS data

Prompt	Response	Description
REQ	TNB	Print TN data.
CUST	xx	Customer number, as defined in LD 15

Determining VMB state

Review the printed VMB data to determine the status of a particular VMB. Valid VMB states appear in Table 85.

Auditing the VMB database

The VMBA application provides both automatic and manual synchronization procedures to help ensure the consistency of the system and Meridian Mail databases. The databases may lose synchronization during one of the following events:

- A technician changes VMBs directly on Meridian Mail, rather than through the system.
- A transaction error occurs during transmission between the system and Meridian Mail.

CAUTION

LD 17 includes a data correction setting (DATA_CORRECT = ON). With this option activated when an audit is run, the system resolves any discrepancy by changing the Meridian Mail database to match the system database. If the databases are out of synchronization because VMB data was changed directly on Meridian Mail, the audit replaces the changed Meridian Mail data with the original system data. Therefore, it is advisable to run an audit initially with DATA_CORRECT = OFF to determine what discrepancies (if any) exist.

Using automatic audit

Responding with ON to the AUTO_AUDIT prompt in LD 17 causes a detailed database consistency check to run every five days. During this audit, Meridian Mail compares its VMB data with each system DN's data. The following are possible results:

- The data for that DN matches.
- Meridian Mail indicates a match to the system.
- The data for that DN does not match, and DATA_CORRECT = ON.
- Meridian Mail changes its data to match the data on the system. A message appears on the system TTY indicating that a discrepancy was detected and corrected.

- The data for that DN does not match, and DATA_CORRECT = OFF.
- A message appears on the system TTY indicating that a discrepancy was detected. Manual intervention is required to correct the discrepancy.

Starting a manual audit

To start the audit function manually, use the ENL VMBA command with the AUDT option in LD 48. The format of the command is as follows:

ENL VMBA <vsid> AUDT <ALL, xxxx>

where

<vsid> is the VAS ID on which the application is configured
ALL specifies that all configured VMBs be audited
xxxx specifies the DN whose VMB is to be audited

Disabling audit

Use the DIS VMBA with the AUDT option to disable the audit function. The format of the command is as follows:

DIS VMBA <vsid> AUDT

where <vsid> is the VAS ID.

This command disables both automatic and manual audits.

Determining audit status

Use the STAT VMBA with the AUDT option to determine the status of an audit. The format of the command is as follows:

STAT VMBA <vsid> AUDT

where <vsid> is the VAS ID.

Output from this command takes the following format:

AUDIT ACTIVE
x AUDITED
y MISMATCHES FOUND/CORRECTED
z ERRORS

where

x is the number of VMBs audited
y is the number of mismatches found (and corrected, if DATA_CORRECT = ON)
z is the number of failed audit operations

Uploading the Meridian Mail VMB database

Existing sites installing the VMBA application may already have VMBs configured on Meridian Mail. To eliminate the need for a technician to add each VMB manually on the system, the VMBA application includes the ability to upload the Meridian Mail VMB database to the system.

The VMB upload command in LD 48 causes the following processing, if the ALL option is specified. The processing is applied to all SCR, SCN, MCR, and MCN DNs configured on the system.

1 For each DN on the system, Meridian Mail checks to see if a VMB is currently defined.

2 If a Meridian Mail VMB exists for the DN, the VMB data associated with the DN, including the VMB name, is uploaded to the system. The system uses the uploaded data to create the VMB data and name (or to replace the existing VMB data and name) for that DN.

CAUTION

If the second or third DNs received from Meridian Mail are greater than four digits (or seven digits, if the DN expansion feature is equipped), they are discarded. A subsequent audit with data correction enabled deletes them from Meridian Mail.

3 If a Meridian Mail VMB does not exist for the DN, and if a VMB is currently configured for the DN on the system, the VMB is deleted.

Note: A name currently configured for the DN on the system is not deleted.

Starting a database upload

To start a database upload, use the ENL VMBA command with the UPLD option in LD 48. The format of the command is as follows:

ENL VMBA <vsid> UPLD <ALL,xxxx>

where

<vsid> is the VAS ID on which the application is configured
ALL specifies that data for all configured VMBs is to be uploaded
xxxx specifies the DN whose VMB data is to be uploaded

Disabling a database upload

Use the DIS VMBA with the UPLD option to disable the upload. The format of the command is as follows:

DIS VMBA <vsid> UPLD

where <vsid> is the VAS ID.

Determining upload status

Use the STAT VMBA with the UPLD option to determine the status of an upload. The format of the command is as follows:

STAT VMBA <vsid> UPLD

where <vsid> is the VAS ID.

Output from this command takes the following format:

UPLOAD ACTIVE
x UPLOADED
y DELETED
z ERRORS

where

x is the number of VMBs uploaded
y is the number of VMBs deleted
z is the number of failed upload operations

Message Intercept

Contents

This section contains information on the following topics:

Feature description 971
Operating parameters 973
Feature interactions 973
Feature packaging 974
Feature implementation 974
Feature operation 976

Feature description

The Message Intercept feature provides an optional recorded announcement when the following features are used:

- Call Forward Status Notification: the user of an analog (500/2500 type) telephone, or Meridian 1 proprietary telephone, going off-hook, receives a recorded message if Call Forward All Calls is activated on the set indicating that the feature is activated.
- Call Park/Off-hook Queuing: the user of an analog (500/2500 type) telephone, or Meridian 1 proprietary telephone, on hold during a call park or while in an off-hook queue, receives a recorded message indicating the condition.

- Ring Again: if Ring Again has been applied to an analog (500/2500 type) telephone, or Meridian 1 proprietary telephone, either locally or remotely, the user of the set receives a recorded message when going off-hook indicating that the feature has been activated.
- Replacement of Confirmation Tone: an analog (500/2500 type) telephone user receives a recorded message, instead of a confirmation tone, indicating the successful activation, by a Flexible Feature Code (FFC), of Permanent Hold.
- Ring Again Activate. A 2500 telephone user receives a recorded message, instead of a confirmation tone, indicating the successful activation, by an FFC, of:
  - Call Forward Activate
  - Call Forward Verify
  - Ring Again Verify
  - Automatic Wake-up Activate
  - Automatic Wake-up Verify
  - Speed Call Store
  - Speed Call Erase
  - Store Number
- Apply Ring Again: an analog (500/2500 type) telephone user, after activating Ring Again, receives a recorded announcement indicating that the feature has been activated.
- Message Waiting: an analog (500/2500 type) telephone, or Meridian 1 proprietary telephone user that has a message waiting receives a recorded announcement when going off-hook indicating that a message is waiting.

- Do Not Disturb: the user of an analog (500/2500 type) telephone, or Meridian 1 proprietary telephone receives a recorded announcement, if the set is under individual or group Do Not Disturb, indicating the condition.
- Set Status Lockout: the user of an analog (500/2500 type) telephone, or Meridian 1 proprietary telephone receives a recorded announcement, if the set is in a lockout state due to Electronic Lockout or Scheduled Access Restriction, indicating the condition.

These messages are provided either by a KAPSCH digital announcer which interfaces externally to the system by way of the QPC605 TDS card, or by other announcers interfaced through the Music Interface. The announcement will continue until either time out or the telephone goes on-hook.

Operating parameters

Message Intercept must be specified on a telephone basis.

Feature interactions

Announcements

Announcements are not available to attendants or trunks.

The actual announcement received by the telephone will be the one defined by the SRC number on the announcer.

It is possible for a connection to the announcer to occur at some point other than the beginning of the message. The message will continue indefinitely, until the call status changes.

A maximum of 30 telephones can be fed announcement at any given time by a single TDS.

Network Drop Back Busy and Off-hook Queuing

If the Message Intercept feature is equipped, a caller in an off-hook queue may receive the message intercept voice response rather than the off-hook queuing tone.

Feature packaging

Message Intercept (MINT) is package 163, which requires:

- Flexible Tone and Cadences (FTC) package 125.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 56 – Configure Message Intercept tones.

2 LD 10 – Allow Message Interception on an analog (500/2500 type) telephone.

3 LD 11 – Allow message intercept on a Meridian 1 proprietary telephone.

LD 56 – Configure Message Intercept tones.

Prompt	Response	Description
...		
MINT	(NO) YES	(Do not) allow tones or announcements.
- CFSN	0-255 0-255	Call Forward All Calls active. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.

- - TDSH	i bb c tt	Tone definition for systems equipped with Tone and Digit cards, where: i = internal (0), or external (1) source bb = burst cc = cadence, and tt = frequency. Prompts with the response i bb c tt define the internal/ external source, burst, cadence and frequency/level respectively. Enter the decimal equivalent (0-15) of the TDS Hex code. The first field is usually 0. If an external source is used, the entry is 1 and the fourth field is 0-7 for the specified channel.
- CPOQ	0-255 0-255	Call is being parked or set is in the off-hook queuing state. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.
- - TDSH	i bb c tt	See above.
- RGAR	0-255 0-255	Ring Again is applied by another set. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.
- - TDSH	i bb c tt	See above.
- RPCT	0-255 0-255	Confirmation Tone replaced by an announcement. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.
- - TDSH	i bb c tt	See above.
- RGAB	0-255 0-255	Station Dialed Busy (calling party allowed to apply Ring Again). The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.
- - TDSH	i bb c tt	See above.
- MWAN	0-255 0-255	Message Waiting. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.

- - TDSH	i bb c tt	See above.
- DNDA	0-255 0-255	Do Not Disturb. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.
- - TDSH	i bb c tt	See above.
- SSLK	0-255 0-255	Set Status Lockout. The first parameter is the MCAD table cadence entry number. The second parameter is the XCT tone code.
- - TDSH	i bb c tt	See above.

LD 10 – Allow Message Interception on an analog (500/2500 type) telephone.

Prompt	Response	Description
...		
CLS	(MIND) MINA	(Deny) allow Message Intercept. You must respond with one of these prompts for each analog (500/2500 type) telephone.

LD 11 – Allow message intercept on a Meridian 1 proprietary telephone.

Prompt	Response	Description
...		
CLS	(MIND) MINA	(Deny) allow Message Intercept. You must respond with one of these prompts for each Meridian 1 proprietary telephone.

Feature operation

The message will continue until it times out, or the telephone is placed on-hook.

Message Registration

Contents

This section contains information on the following topics:

Feature description 977
Operating parameters 977
Feature interactions 979
Feature packaging 981
Feature implementation 981
Feature operation 983

Feature description

Message Registration (MR) allows customers to meter local calls so that Hospitality administration can read, change, and reset message units stored on the meters.

Software meters accumulate call charges for room phones, administration phones, customer phones, Attendant Consoles, incoming TIE trunks, and Central Office (CO) trunks.

Operating parameters

Meters are incremented when Reverse Battery (RVB) signals are received from loop start or ground start Central Office (CO) trunks. The meter is incremented once for each completed local call, regardless of duration, against the originating Directory Number (DN). No charge is made to any meter if a call over a metered route is not established.

Metering is applied on a route basis. When provisioning a customer for the MR feature, calls that are to be metered can have access only to routes that are metered. Metered calls cannot be overflowed to a nonmetered route.

One software meter is assigned to every telephone Directory Number, attendant DN, and Trunk Access Code (TRC) that requires metering. Each software meter can count up to 32,766 calls before being automatically reset to zero. Prior to reset, the meter contents are displayed on the system background terminal.

The ATTN meter accumulates charges for all metered calls made by Attendant Consoles within a customer group. The TRK meter is provided for each incoming tie trunk route and Central Office (CO) route. Charges are registered for tandem call connections made by incoming TIE trunks over a meter-assigned route. One overflow meter, the CUST meter, allows each customer to accumulate any charges that cannot be registered to another meter.

With call modification, the party originating the metered call has its meter charged. Once the meter is charged, the charge cannot be transferred to another party's meter through Call Modification.

Attendant-originated calls to metered routes are charged to the party connected to the call source. If no party is connected to the source, the attendant's meter is charged.

If the attendant originates a call to a CO trunk, and the call is not extended to an internal Directory Number, the attendant's meter is incremented.

Incoming TIE trunks involved in metered tandem calls are charged to a meter associated with the route, to allow for billing to a party other than the customer.

Metered calls made within the customer that cannot be charged to any other meter are charged to the overflow meter associated with the CUST meter.

Message Registration (MR) uses only the Reverse Battery (RVB) type of answer supervision. Periodic Pulse Metering is not supported.

A QPC219, QPC330, or QPC450 trunk card must be used for the CO trunk routes receiving Reverse Battery Signals (RVB). Also, a QPC330 card must have its signaling set up as for a QPC219 trunk card.

The NT8D14 Universal trunk does not provide MR.

A Background Terminal (BGD) assigned meter access Controlled Class of Service (CCOS) can automatically read, change, or print meter values. The reading, changing, and printing can also be done manually. From a BGD, any meter can be turned on or off (for instance, set to accumulate or not accumulate charges), except for the customer meter, which is always on. When the BGD accesses a meter, a classification indicating the meter type is shown. The five possible meter classifications are:

- ROOM (room number)
- ADMN (administration)
- ATTN (attendant console)
- TRK (trunk)
- CUST (customer/miscellaneous)

For detailed information regarding Background Terminal (BGD) commands for MR, refer to *Background Terminal User Guide*.

Meter contents can also be read or changed by a Meridian 1 proprietary telephone equipped with a Message Registration key/lamp pair (MRK) and a display. The M2317 telephone can also be used. Three values are shown on the display for MR:

- the Directory Number (DN) of the telephone whose meter value is being changed
- the existing value of the meter
- the new value being entered

An MRK cannot be assigned to Automatic Call Distribution (ACD) agents.

The Call Detail Recording (CDR) feature does not display message registration meter information.

Feature interactions

Attendant Administration

MR service change is not supported by Attendant Administration.

Call Forward All Calls
Call Transfer
Conference

The party that originates a call is charged. The charge cannot be moved to another party using Transfer, Conference, or Call Forward All Calls.

Coordinated Dialing Plan
Centralized Attendant Service

MR is mutually exclusive of Coordinated Dialing Plan and Centralized Attendant Service.

Maintenance

Any maintenance testing done on metered trunks does not affect the meter values.

Multiple Appearance Directory Number

For Multiple Appearance Directory Number (MADN), the system selects the appropriate meter for the DN based on following this procedure:

1. It accesses the meter of the most recently configured telephone having a Prime DN (PDN) appearance and Message Registration Allowed (MRA) Class of Service.
2. If no Terminal Number (TN) in the DN block has MRA Class of Service, the customer meter is charged. For the Message Registration Key (MRK), the system provides overflow and sets the MRK lamp to flash. For the Background Terminal (BGD), it prints a NO DATA FOUND message.

Multi-Tenant Service

The ability to retrieve or update hotel or motel Room Status (RMS) and meter counts exists at the customer level, not at the tenant level.

Trunk to Trunk Connection

The last party releasing the call collects the total value of outstanding Periodic Pulse Metering (PPM) generated on outgoing trunks. If the last party is an internal set, the outstanding PPM is stored against the meter of the set. If the last party is an internal TIE trunk, the outstanding PPM is stored against the meter associated with the internal TIE trunk access code. If the last party is an outgoing external trunk, the outstanding PPM is stored against the meter associated with the external trunk access code.

Feature packaging

Message Registration (MR) package 101 requires:

- Controlled Class of Service (CCOS) package 81, and
- Background Terminal Facility (BGD) package 99.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Activate Message Registration on routes.

2 LD 14 – Configure the polarity trunk.

3 LD 10 – Allow or deny analog (500/2500 type) telephones access to meters.

4 LD 11 – Allow or deny Meridian 1 proprietary telephones access to meters.

LD 16 – Activate Message Registration on routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.

CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	aaa	Trunk route type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, FGOT, ISA, MCU, MDM, MUS, PAG, R232, R422, RAN, RCD, RLM, RLR, or TIE.
- MR	(NO) YES RVB	Only prompted if TKTP = COT or FGOT; MR provided on (no routes), all routes, or Reverse Battery (RVB) routes.

LD 14 – Configure the polarity trunk.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	COT	CO trunks.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(PIP) PSP	Polarity (insensitive) or sensitive. Use PSP for QPC218, QPC219, QPC295. Use PIP for QPC330, QPC331.

LD 10 – Allow or deny analog (500/2500 type) telephones access to meters.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(MRD) MRA	MR (denied) or allowed.

LD 11 – Allow or deny Meridian 1 proprietary telephones access to meters.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, or 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	aaa	Digit Display options, where: aaa = ADD, DDS, NDD.
	(MRD) MRA	MR (denied) allowed.
KEY	xx MRK	MR key, where: xx = key number.

Feature operation

No specific operating procedures are required to use this feature.

Message Waiting Indicator by Directory Number

Contents

This section contains information on the following topics:

Feature description 985
Operating parameters 992
Feature interactions 992
Feature packaging 993
Feature implementation 994
Feature operation 1001

Feature description

The Message Waiting Indicator by Directory Number (MWDN) feature increases the flexibility in presenting a message waiting indication on the M2006, M2008, M2016, M2216, and M2616 proprietary sets. The MWDN feature provides the following functionalities:

- presentation of multiple message waiting indications on one set
- presentation of multiple message waiting indications for one mailbox on more than one set
- presentation of remote message waiting indications for message monitoring
- support for one mailbox for multiple Directory Numbers (DNs)

Multiple message waiting indications on one set

Prior to the MWDN feature, where more than one DN was configured on one set, only the Primary Directory Number (PDN) -- or the single appearance non-PDN -- had a Message Waiting Key (MWK) and the LED for the message waiting indication. There was no message waiting indication for DNs other than the PDN.

The MWDN feature allows a user to have a separate MWK, called the Extended Message Waiting Key (XMWK), for each of the mailbox DNs configured on that set. The DN associated with the XMWK must be configured as a non-PDN on that set.

The XMWK starts flashing when a new voice message is received for the DN associated with this key. Once all the new voice messages have been retrieved, the indication on the XMWK associated with that DN is canceled.

Multiple message waiting indications on one set has application for environments where one set has the DNs for several individuals. Figure 66 shows a scenario where an administrative assistant monitors the DNs for several individuals from the set.

Multiple message waiting indications for one mailbox on more than one set

Prior to the MWDN feature, if there was more than one appearance of a DN, the MWK could be turned on or off only for the primary appearance of that DN. With the MWDN feature, when a mailbox DN appears on more than one set, the XMWK can be configured for the non-primary appearance of the mailbox DN on each set. The DN associated with the common mailbox must be configured as a non-PDN on all the sets where it appears (except for the one PDN set).

When a new voice message is received for the DN associated with the common mailbox, all the XMWKs configured on all the sets and associated with this DN start flashing. Once all the new messages from the common mailbox have been retrieved by any of the users, the message waiting indication on all the XMWKs associated with the general mailbox DN is canceled.

Figure 66
Multiple message waiting indications on one set

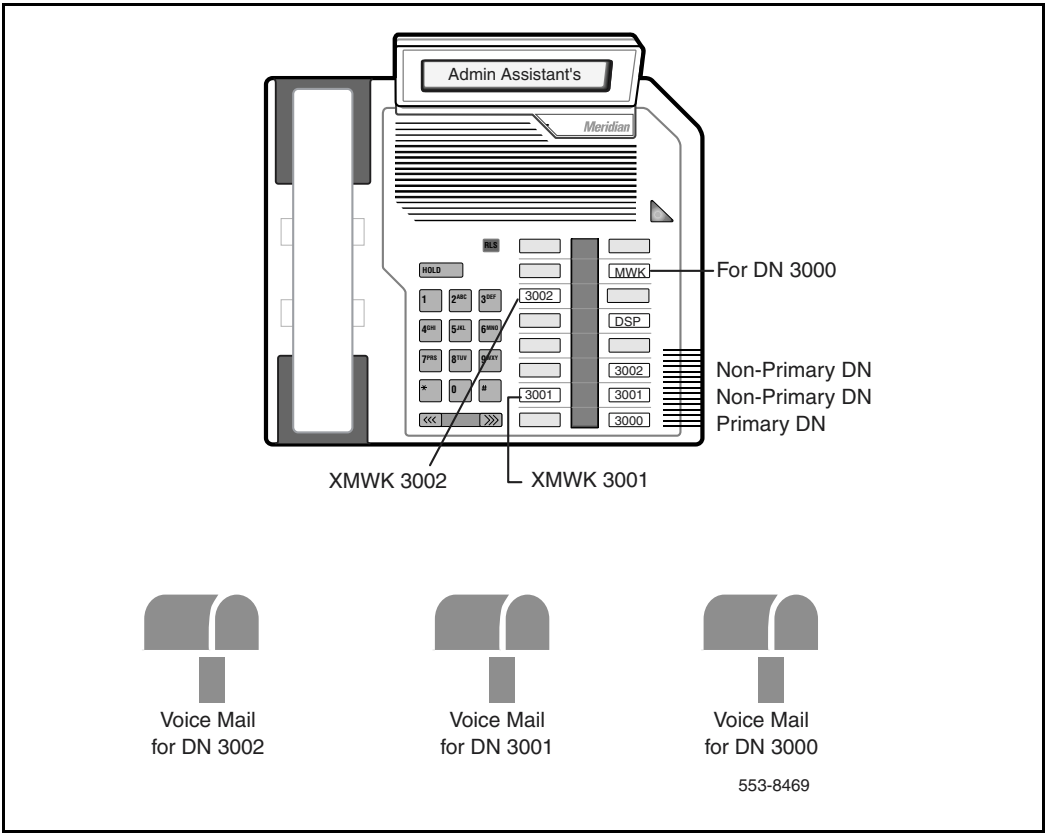
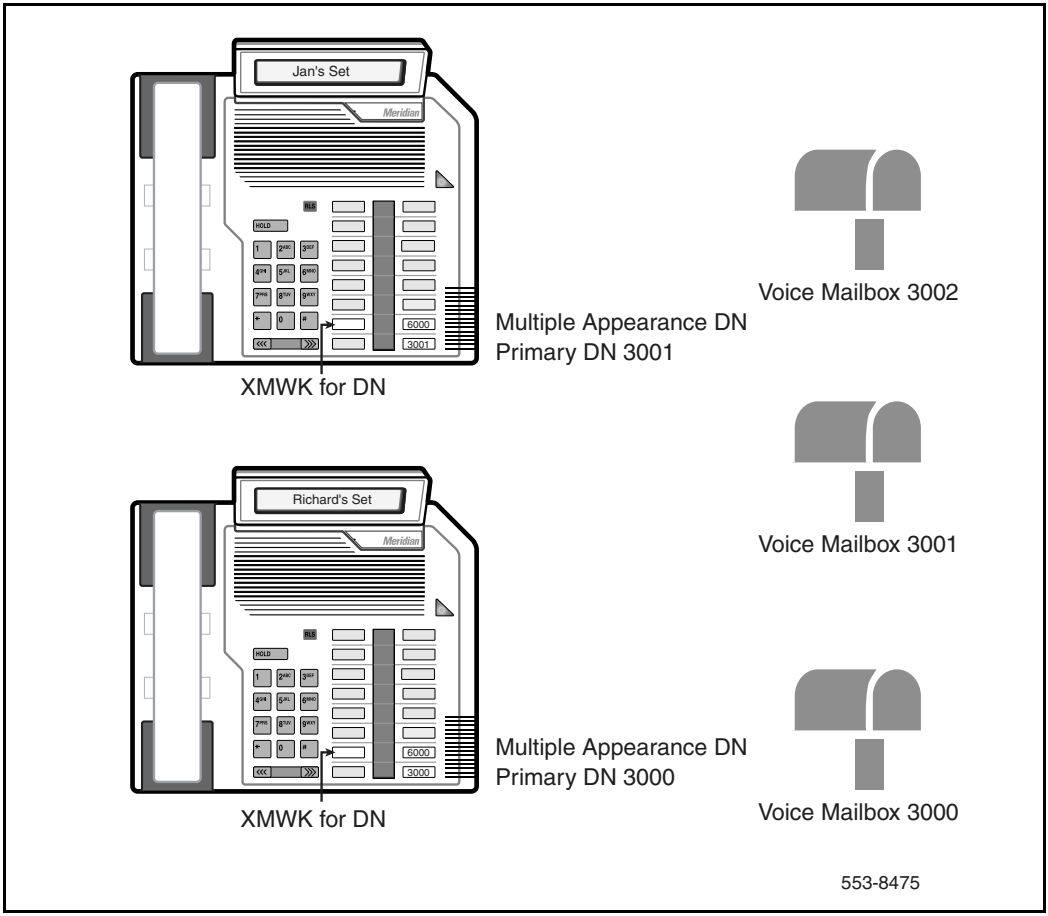


Figure 67 shows multiple message waiting indications for one mailbox on many sets. An application of this component of the MWDN feature is a person with more than one set with a shared DN, such as a mobility user in a macrocellular environment (that is, within a building). With the MWDN feature, messages coming into this DN will light up the message waiting indicators both their desk set and their mobility telephone. In this scenario, both the desk set and mobility telephone must be on the same switch.

Figure 67
Multiple message waiting indications for one mailbox on more than one set



Remote message waiting indication for message monitoring

Prior to the MWDN feature, if a new voice message was received, users had to see the message waiting indication or log in to their mailbox from a remote set to query if they had voicemail. With the MWDN feature, users can monitor the status of their mailboxes from a remote set without logging into their sets.When a new message arrives to the monitored mailbox DN, the message waiting indication is propagated to the Remote Message Waiting Key (RMWK) on a remote set that is programmed for that mailbox DN. The RMWK monitors those DNs which have at least one primary appearance.

The Message Center DN must be configured; configuring the monitored mailbox DN is optional. The RMWK for the mailbox DN is user programmable from the set.

When programmed, the RMWK starts flashing if any new voice message arrives for the associated mailbox DN; if not, the RMWK remains steadily lit. To cancel the RMWK function, press the RMWK when it is lit or flashing.

The temporary redirection and message waiting indicator propagation of a Phantom TN uses this component of the MWDN feature. Figure 68 illustrates a Phantom TN, DN 3001, with a RMMA/RMMO class of service. A RWMK key is configured on a set to monitor the messages for the DN 3001. Any new voice message to the Phantom DN 3001 is shown on the RMWK. When a new voice message is received for DN 3001, the RMWK starts flashing; once all the new voice messages are retrieved for DN 3001, the RMWK becomes steady lit.

One mailbox for multiple DNs

Prior to the MWDN feature, three DNs could be associated with one mailbox; however, only the PDN which shares the mailbox displayed the message waiting indication. The MWDN feature extends the message waiting indication to all proprietary set appearances on which the three DNs sharing the mailbox are configured.

This feature is used in an environment such as a technical support area with up to three technicians having their own DN but sharing a common mailbox. Figure 69 shows DNs 5000, 5001 and 5002 with a shared mailbox.

Figure 68
RMWK operation when a Phantom TN is call forwarded to a set

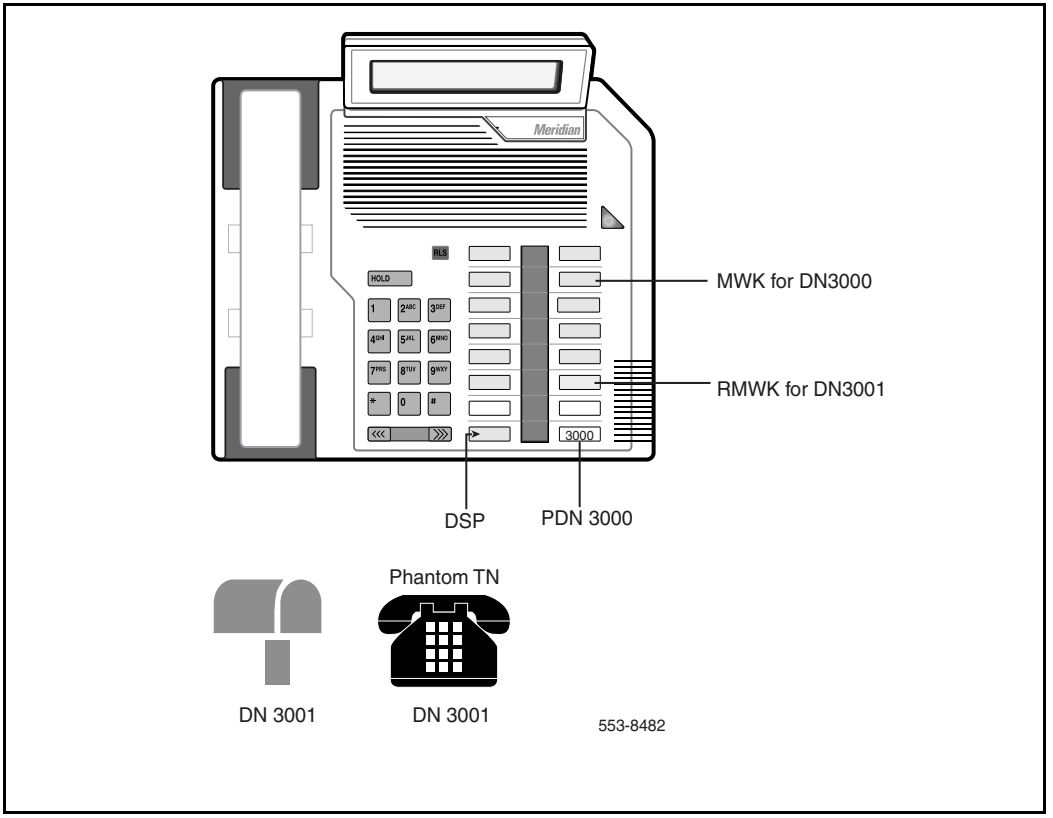
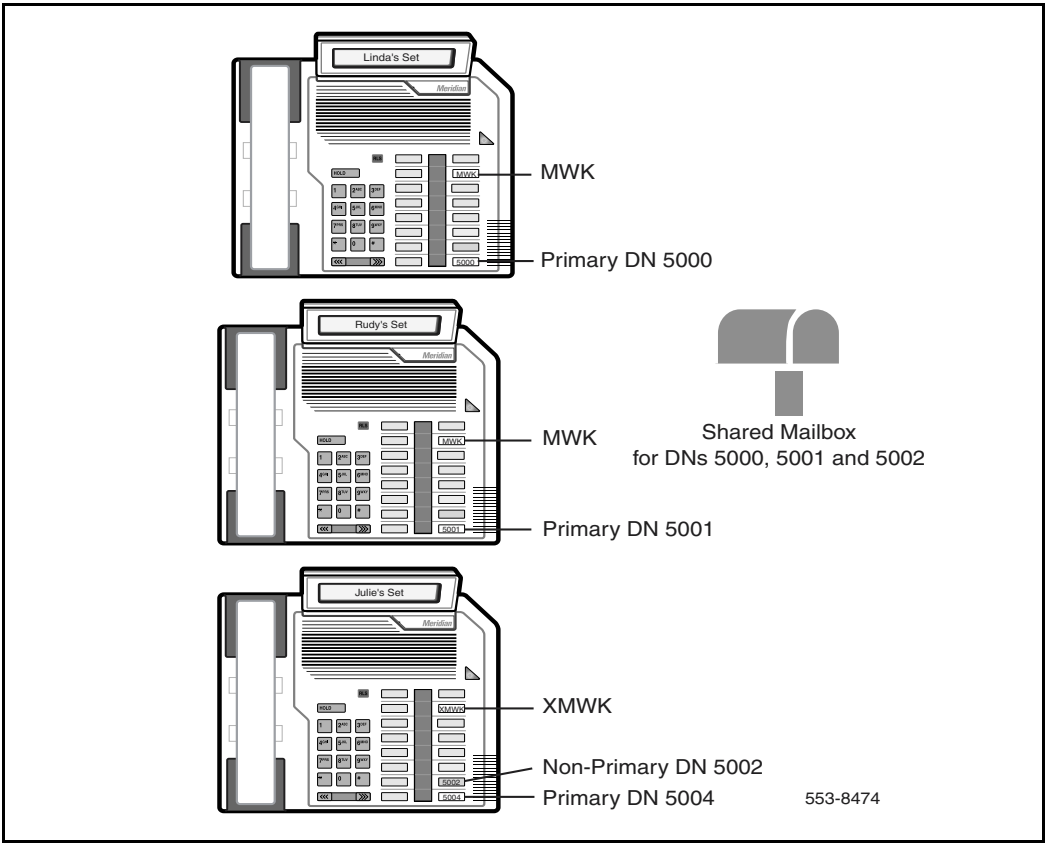


Figure 69
One mailbox for multiple DNs



Operating parameters

MWDN supports features within the same node; it does not support features on different nodes across a network. For example, MWDN supports Meridian Mail if it is on the same node; the MWDN feature does not support Meridian Customer Defined Network (MCDN) messaging services across a network.

Meridian Mail 9 is required to support one mailbox for multiple DN functionality. The Voice Mailbox Administration (VMBA) package 246 must be equipped to enable the functionality of one mailbox for multiple DNs.

The MWDN feature does not support message waiting indication in the macrocellular environment.

The Remote Message Waiting Key (RMWK) monitors PDNs only.

A DN can be monitored by only one RMWK at a time.

Each Extended Message Waiting Key (XMWK) can be associated with one non-PDN only on each set.

Feature interactions

Display key

With the MWDN feature, the Display key (DSP) shows the Message Center DN and the mailbox DN associated with the XMWK and the RMWK. This display occurs when a user presses the DSP and then either the XMWK or the RMWK. If there is no mailbox DN associated with the RMWK, only the Message Center DN is displayed.

Multiple Appearance DN

For the Multiple Appearance DN feature:

- On sets where the DN is configured as a PDN, the message waiting indication occurs on the MWK and red LED.
- On sets where the DN is configured as a non-PDN, the message waiting indication occurs on the XMWK and the red LED depending upon the LMPN or LMPX class of service. The LMPN class of service is defined as the red LED reflects the status of the mailbox associated with the PDN. The LMPX class of service is defined as the red LED reflects the status of the mailboxes associated with both PDN and non-PDNs.

The RMWK can be used to monitor a Multiple Appearance DN if the DN has at least one primary appearance.

Phantom Terminal Number

The Phantom Terminal Number feature permits users to define and configure Terminal Numbers (TNs) with no associated physical hardware. The Phantom TN can be associated temporarily with a physical set. With the MWDN feature, a user can monitor the mailbox associated with the Phantom DN through the RWMK on a proprietary set.

Feature packaging

The MWDN feature requires these packages:

- Digit Display (DDSP) package 19
- Message Waiting Lamp Maintenance (MWC) package 46
- Voice Mailbox Administration (VMBA) package 246

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable the Message Center in the Customer Data Block.

2 LD 11 – Configure the class of service option for the Extended Message Waiting Key (XMWK) and its LEDs on the proprietary sets.

3 LD 11 – Configure the Remote Message Waiting Key (RMWK) to monitor remote sets with a mailbox.

4 LD 10 – Configure new class of service options to enable analog (500/2500) sets to be monitored remotely.

5 LD 11 – Configure the class of service to enable Meridian 1 proprietary sets to be monitored remotely.

6 LD 10 – Extend the message waiting indication function to all the analog (500/2500) sets on which the DNs sharing the mailbox are configured.

7 LD 11 – Extend the message waiting indication function to all the Meridian 1 proprietary sets on which the DNs sharing the mailbox are configured (whether the DNs are PDN or non-PDN).

For all the following tasks, first enable the Message Center in the Customer Data Block in LD 15. See page 995.

To configure multiple message waiting indications on one set or on many sets:

- Configure the class of service options for the Extended Message Waiting Key (XMWK) and its LCDs on proprietary sets in LD 11. See page 996.

To configure remote message waiting indications on one set:

- Configure the Remote Message Waiting Key (RMWK) to monitor remote sets with a mailbox in LD 11. See page 997.
- Configure new class of service options to enable analog (500/2500) sets to be monitored remotely in LD 10. See page 998.

  Note: All sets with primary DNs to be monitored **must** be configured as RMMA/RMMO.

- Configure new class of service options to enable proprietary sets to be monitored remotely in LD 11. See page 999.

  Note: All sets with primary DNs to be monitored **must** be configured as RMMA/RMMO.

To configure message waiting indications on sets where DNs sharing a mailbox appear:

- Configure the class of service option to enable the message waiting indication for all the analog (500/2500) sets on which the DNs sharing the mailbox are configured in LD 10. See page 1000.
- Extend the message waiting indication function to all the Meridian 1 proprietary sets on which the DNs sharing the mailbox are configured in LD 11. See page 1001.

LD 15 – Enable the Message Center in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	RDR	Call Redirection.
CUST	xx	Customer number, as defined in LD 15
OPT		Options.
	MCI	Message Center Included.

To configure the XMWK key, the DN to be associated with the XMWK must be configured as a non-Primary Directory Number (non-PDN) on this set.

Note: The DN associated with the XMWK must not have an XMWK already associated with it on this set.

LD 11 – Configure the class of service option for the Extended Message Waiting Key (XMWK) and its LEDs on the proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied.
	LMPX	The red LED on the proprietary sets reflects the status of the mailbox associated with both the PDNs and non-PDNs with the associated Extended Message Waiting Keys (XMWKs) or the Remote Message Waiting Keys (RMWKs). (LMPN) = The red LED on proprietary sets reflect the status of the mailbox associated with the PDNs.
...		
KEY		Telephone function key assignments.
	xx XMWK xxxx yyyy	Extended Message Waiting indication key Where: xx = key number xxxx = Message Center DN yyyy = mailbox DN ***Note:*** XWMK cannot be configured on key 0.

LD 11 – Configure the Remote Message Waiting Key (RMWK) to monitor remote sets with a mailbox.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied.
	LMPX	The red LED on the proprietary sets reflects the status of the mailbox associated with both the PDNs and non-PDNs with the associated Extended Message Waiting Keys (XMWKs) or the Remote Message Waiting Keys (RMWKs). (LMPN) = The red LED on proprietary sets reflect the status of the mailbox associated with the PDNs.
...		
KEY		Telephone function key assignments.
	xx RMWK xxxx [yyyy]	Remote Message Waiting indication key Where: xx = key number xxxx = Message Center DN [yyyy] = DN to be monitored {optional}

Note: All sets with primary DNs to be monitored must be configured as RMMA/RMMO.

LD 10 – Configure new class of service options to enable analog (500/2500) sets to be monitored remotely.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	500	Analog (500/2500) set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	RMMA RMMO	Allow the set to be remotely monitored for messages. Allow the set to be remotely monitored for messages and allow the set to override, if it is being monitored already. (RMMD) = Deny set for Remote Monitoring of Messages.

Note: All sets with primary DNs to be monitored must be configured as RMMA/RMMO

LD 11 – Configure the class of service to enable Meridian 1 proprietary sets to be monitored remotely.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	LMPX	Enable the red LED on the supported proprietary sets to reflect the status of the mailboxes associated with both PDN and non-PDNs. (LMPN) = do not enable the red LED on the supported proprietary sets to reflect the status of the mailboxes associated with both PDN and non-PDNs.
	RMMA RMMO	Allow set for Remote Monitoring of Messages. Allow set for Remote Monitoring of Messages and Override, if it is being monitored already. (RMMD) = Deny set for Remote Monitoring of Messages.

Note: Voice Mailbox Administration (VMBA) must be configured before configuring one mailbox supporting Multiple Appearance DNs. Refer to *Hospitality Features: Description and Operation* (553-3001-353) for information on configuring VMBA.

LD 10 – Extend the message waiting indication function to all the analog (500/2500) sets on which the DNs sharing the mailbox are configured.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	500	Analog (500/2500) set.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	SMWA	Allow Extended Message Waiting Indication. (SMWD) = Deny Extended Message Waiting Indication.

Note: Voice Mailbox Administration (VMBA) must be configured before configuring one mailbox supporting Multiple Appearance DNs. Refer to *Hospitality Features: Description and Operation* (553-3001-353) for information on configuring VMBA.

LD 11 – Extend the message waiting indication function to all the Meridian 1 proprietary sets on which the DNs sharing the mailbox are configured (whether the DNs are PDN or non-PDN).

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	aaaa	Type of set. aaaa = 2006, 2008, 2016, 2216, 2616.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed.
	SMWA	Allow Extended Message Waiting Indication. (SMWD) = Deny Extended Message Waiting Indication.

Feature operation

Remote message monitoring:

1. Press the RMWK with the set in idle position.
2. The set winks and displays RMWK XXXX (where XXXX is the existing mailbox DN) prompting for a new mailbox DN. If there is no mailbox DN, the set displays RMWK.
3. Enter the new mailbox DN.
4. The screen displays the digits. Press the RMWK to validate the mailbox DN.

   Note: If you press the RMWK without entering the digits, the RMWK remains programmed for the DN which was stored previously. If there is no DN stored, the Overflow tone is given.

5 If the DN is invalid, the Overflow tone is given. If the mailbox DN is valid:

— If the set on which this DN is configured as PDN has a class of service set to RMMA or RMMO and is not monitored:

– the RMWK starts flashing if there are any new voice messages for this DN.

– the RMWK lamp becomes steady lit and the screen changes to idle mode if no new voice message exists for this DN.

— If the set on which this DN is configured as PDN has a class of service set to RMMO and is being monitored by another set, this set overrides and continues to monitor.

— Overflow tone is given if any set on which this DN is configured as PDN has class of service set to RMMD.

— Overflow tone is given if any set on which this DN is configured as a PDN has class of service set to RMMA and is being monitored by another set.

6 To cancel remote message monitoring, press the RMWK when it is lit or flashing.

Message Waiting Lamp Maintenance

Content

The following are the topics in this section:

Feature description 1003
Operating parameters 1004
Feature interactions 1004
Feature packaging 1004
Feature implementation 1004
Feature operation 1005

Feature description

This maintenance enhancement alleviates the “dark effect” when neon lights are tested in low ambient light conditions.

Because the dark effect is inherent to neon lamps, it is recommended that PBXT Message Waiting Lamp tests not be run during low ambient light conditions. The line card detector circuitry can register lamp failures under these circumstances, and the Message Waiting Lamp test may be unreliable. Lamps are listed as faulty when they fail the test once in three attempts.

The PBXT Message Waiting Lamp tests can be run under one of the following conditions:

- automatically at a system-specified time, or
- manually at any time (LD 32).

Automatic scheduling should consider low traffic times, when there is still enough ambient light to avoid the dark effect. To prevent the automatic scheduling of LD 32, LD 32 must be excluded from the daily routines ("midnights") and the system-defined hour must be the default "X" value.

When the hour defined defaults to the "X" value, an error message is output to remind the customer that the PBXT tests are still part of the daily routines, unless LD 32 is removed from the list.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The Message Waiting Lamp Maintenance feature requires Message Waiting Center (MWC) package 46.

Feature implementation

LD 17 – Define the time for the maintenance tests.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	OVLY	Gate opener.
OVLY	(NO), YES	Change overlay area options.
- PBXH	hh	PBX Hour for maintenance tests, where: hh = hour for tests, 0-23.
	x	Enter x if no tests are to be performed.

Feature operation

No specific operating procedures are required to use this feature.

Message Waiting Unconditional

Contents

This section contains information on the following topics:

Feature description 1007
Operating parameters 1007
Feature interactions 1008
Feature packaging 1011
Feature implementation 1011
Feature operation 1011

Feature description

This feature enhances the use of the Message Indication key (MIK) and Message Cancellation key (MCK) by an Automatic Call Distribution (ACD) message center agent or message center attendant.

This feature enhancement applies to a Network Message Center. It is configured on a customer basis in LD 15.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

ACD Message Center

The operation of ACD Message Center telephones is basically the same as an ACD system with incoming call queues and available agent queues. The ACD Message Center cannot operate in combination with an Attendant Message Center. However, if all telephones are in the Make Busy mode (not logged in), Message Center calls can be routed to the attendants who can then function as the message center. Queue overflow features are allowed for a Message Center ACD DN in the same way as for any other ACD system with the properly equipped package. Other ACD features, such as RAN and Music, operate as for a normal ACD system with the appropriate packages.

A Message Center operator cannot originate calls on the MSG IN-CALLS key; therefore originating features are not applicable on this key. Separate DN keys must be provided for these functions.

DN Message Center

The Message Center DN must be the prime DN, otherwise all normal features can be assigned to this DN.

Attendant Message Center

Once a call is extended to an ACD Message Center by an attendant, it is released completely from attendant operation, and features, such as recall and camp-on, cannot be activated. For calls extended to a DN Message Center, normal attendant functions, such as recall and camp-on, can be used. Other attendant functions operate normally.

Call Forward (All Calls)

Call Forward should be denied at telephones serving as the message center. On a telephone basis, Call Forward takes precedence over the message center. If a call is forwarded to another telephone, activation of message waiting depends on whether or not the second telephone has message waiting allowed.

Call Forward Message Waiting dialtone can be provided to 500/2500 type telephones. This is an indication that Call Forward All Calls is active and a message is waiting at the message center.

Call Forward, Internal Calls

The Message Center treats Internal CFW in the same way as Call Forward All Calls (CFAC).

Call Forward Message Waiting dialtone can be provided to 500/2500 type telephones as an indication that Call Forward, Internal Calls is active and a message is waiting at the message center.

Call Forward Busy

Call Forward Busy (CFB) should be denied at telephones serving as the message center. An option is provided to allow DID calls to a busy telephone to be routed to the message center. If this option is selected by the customer, message waiting takes precedence over the customer-defined path for CFB.

Call Forward No Answer

Call Forward No Answer (CFNA) should be denied at telephones serving as the message center. On a telephone user basis, message waiting takes precedence over the customer defined path of CFNA.

The capability to light and extinguish message waiting lamps can be used in conjunction with CFNA to simulate a multiple message center. Any telephone equipped with message lamps, but without message waiting allowed class of service, can CFNA to specified DNs on the telephones equipped with MSG INDIC and MSG CANC key/lamp pairs.

These telephones have the capability to light or extinguish message waiting lamps by manually entering the DN of the telephone for which a message was taken. Call processing is the normal call processing for CFNA, not the message center call processing. When a call is forwarded, the MSG INDIC lamp does not light since this is not true message center operation.

Call Transfer/Conference from an Analog (500/2500 type) telephone

Message waiting interrupted dial tone is not provided when the user flashes the switchback to activate Call Transfer or Conference. The normal dial tone for this purpose is provided.

Flexible Call Forward No Answer to any DN

Flexible Call Forward No Answer (CFNA) to any DN forwards unanswered calls to a pre-designated CFNA DN. All telephones with message waiting allowed have the CFNA DN assigned to the message center regardless of whether Flexible CFNA has been selected by the customer or whether CFNA is allowed or denied for the telephone.

Hunting

Hunting should be denied at telephones serving as the message center (MC). On a user basis, hunting takes precedence over message waiting. However, message waiting can be activated after hunting provided the hunted telephone is message waiting allowed and does not answer the call. If desired, the MC DN can be specified as the hunt number.

Listed Directory Number

A message center can be assigned to a Listed Directory Number (LDN) and behaves in a similar manner to an attendant message center. The calls come in on an LDN ICI instead of the MSG CENTER ICI, and direct message calls do not activate the MSG CANC key. The operator must access the user telephone directly to cancel that telephone's message indication.

Ring Again for an Analog (500/2500 type) telephones

Message waiting interrupted dial tone is not provided when the user flashes the switch back to activate Ring Again. The normal dial tone for this purpose is provided.

User Selectable Call Redirection

User Selectable Call Redirection allows the user to perform two tasks:

- To assign the four redirection DNs from the telephone. These DNs include the CFNA DN and the external CFNA DN (if it exists).
- To change the way the number of ringing cycles are defined for Flexible Call Forward No Answer (CFNA). One of three options can now be selected from the telephone.

This feature does not support Basic Rate Interface (BRI) telephones.

Feature packaging

Message Waiting Center (MWC) package 46.

Feature implementation

LD 15 – Enable the Message Waiting Unconditional feature enhancement for a customer.

Prompt	Response	Description
...		
OPT	(MWUD) MWUA	Message Waiting Unconditional feature enhancement (denied) allowed.

Feature operation

The current operation is such that, if an internal call or an incoming external call to a station is not answered, the caller may leave a message at the message center (ACD agent or message center attendant). To activate or deactivate a message waiting indication on the desired station, the ACD agent or attendant presses the Message Indication key (MIK) and Message Cancellation key (MCK), respectively. To use this method when the message center has an active call, the active call must be placed on hold, or the message center attendant has to be placed in position busy or the ACD agent in Not Ready state before the MIK/MCK may be activated.

The enhanced operation allows the Message Indication key (MIK) and Message Cancellation key (MCK) to be used unconditionally (that is, if there is a call presented to the message center, and not answered, pressing the MIK or MCK takes precedence over the presented call).

> ***Note:*** This enhancement applies only to presented calls which have not been answered. If the message center has a call already established, the current operation applies.

Mixed disk drive sizes with CP PII

Contents

This section contains information on the following topics:

Feature description 1013
Operating parameters 1013
Feature interactions 1014
Feature packaging 1014
Feature implementation 1014
Feature operation 1014

Feature description

This feature provides the functionality of disk redundancy for Multimedia disk units (MMDU) of different sizes on the respective cores. Regardless of the MMDU sizes, this feature configures them to be exactly 6GB. Therefore the existing functionality of disk redundancy is automatically preserved.

This feature uses the configuration option provided by the system software to configure MMDUs of any size to 6GB. This feature is automatically enabled during sysload. There is no manual intervention needed.

Operating parameters

The feature is not applicable for MMDU sizes less than 6GB.

Feature interactions

All existing features are unaffected.

Inventory feature

Since no pack ID is available, there is no data provided to the Inventory Reporting Feature.

Succession 1000M Single Group and Meridian 1 Option 61C CP PII

The Succession 1000M Single Group and Meridian 1 Option 61C CP PII systems use the install disk, which is modified to support the Mixed Disk Drive feature.

Feature packaging

No new software packages are introduced.

Feature implementation

The feature is automatically enabled during sysload. There is no manual intervention needed.

Feature operation

There are no specific procedures required to operate this feature.

Multi-language Messaging

Contents

This section contains information on the following topics:

Feature description 1015
Operating parameters 1016
Feature interactions 1017
Feature packaging 1017
Feature implementation 1017
Feature operation 1018

Feature description

System software has a system of message reporting that issues reports containing English sentences in addition to error codes and data, and the Error Message Lookup feature that enables Nortel Networks technical publication (NTP) explanations of any error code to be displayed on the TTY. The Multi-language Messaging feature enhances these capabilities by providing an additional language besides English. In addition, the capacity to toggle from one language to another without suspending system operations is provided.

Operations, Administration and Maintenance users on Large Systems will now have the ability to have some messages, currently printed in English, to be displayed and logged using another language. The following messages are affected:

- Maintenance and start-up messages specific to Large Systems, with the following exceptions:

  - — Messages printed by the VxWorks OS
  - — Read Only Memory Firmware (ROM F/W) messages
  - — Interactive messages from overlays
- Explanation texts printed by the System Message Lookup Utility (MLU).

Operating parameters

This feature applies to Large Systems.

Every system can support only one language besides English, and changing this language to another language is only possible after re-initializing the system.

Upgrading the messages database to a more recent version is only possible by upgrading the software.

The language is selected for the whole system and changes simultaneously on all configured terminals and log files.

Messages are logged in the Report Processing Tool (RPT) log file as they come (that is, in the language currently configured in LD 17). Hence, the log file may contain messages in both English and the alternate language.

After the language option has been changed in LD 17, some messages may be displayed in the previous language, because they were sent to the printer queue immediately before service change.

At system start, the first messages will be displayed in English, since the current language will not yet have been read from the disk.

It is not possible for the current feature to enable translation of interactive or hard-coded messages in the system.

Translation is not possible for the following messages:

- Large System installation tools screens
- LDs 135 and 137 (interactive messages)
- Application modules

- New/existing tools for database consistency checks
- VxWorks OS
- Messages printed during initialization and SYSLOAD by the system software
- Liquid Crystal Display (LCD) displays on the Central Processing Unit (CPU) board, and
- ROM messages.

No new hardware is required for this feature.

Feature interactions

This feature is an improvement based on the Message Lookup Utility (MLU) and the Report Processing Tool (RPT).

Feature packaging

The following package must be activated for the Multi-language Messaging feature to operate: Multi-language TTY Input/Output (MLIO) package 211.

The following package must be activated to gain access to the System Message Lookup feature: System Message Lookup (SYS_MSG_LKUP) package 245.

Feature implementation

LD 17 – Select which messages to translate.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters
...		
PARM	YES	System parameters.

...		
- NDIS	(NO) YES	New distinctive ringing.
- TRNS		Selects which messages are going to be translated.
	(NONE)	NONE = Help and Large System specific system messages are printed in English.
	HELP	HELP = Help is printed in the translated language and Large System specific system messages are printed in English.
	BOTH	BOTH = Help and Large System specific system messages are printed in the translated language. ***Note:*** The translated language printed is dependent on the software packaging.

Feature operation

No specific operating procedures are required to use this feature.

Multi-Party Operations

Contents

This section contains information on the following topics:

Feature description 1019
Operating parameters 1024
Feature interactions 1024
Feature packaging 1033
Feature implementation 1033
Feature operation 1040

Feature description

Multi-Party Operations (MPO) introduces a number of capabilities. The capabilities are:

Call Join

Allows Meridian 1 proprietary telephone users to conference a held party into an active call without having to redial the held party.

Call Join applies to all Meridian 1 proprietary telephones, regardless of the Class of Service assigned, that are equipped with a Three-party (AO3) or Six-party (AO6) Conference key and at least one secondary DN or Call Waiting key.

This feature allows a Meridian 1 proprietary telephone user to conference a party held with an active party on their set, or transfer the active party to the held party by forming a conference then disconnecting.

Call Join is not available at the Attendant Console.

Three-party Service

Allows analog (500/2500 type) telephone users to toggle between two parties with the option of forming a conference between them, or releasing the active party and reconnecting the held party. Included under the Three-party Service capability are:

- **Three-party Service Timer** – A programmable timer to allow dialing of a Control Digit after a Register Recall.
- **Consultation Call Disconnect Option** – An option to provide alternative treatment to the parties involved in a Consultation call when the Consultation connection is released.

Three-party Service applies to analog (500/2500 type) telephones with Three-party Service Allowed (TSA) Class of Service.

During a normal two-party call, the user can place the established call on hold and originate another call. After the second call is established, the user can:

a. Dial the Conference Control Digit (CNFD) to form a three-party conference between the user, held, and active parties, or transfer the active party to the held party by forming a three-party conference then disconnecting.

b. Dial the Toggle Control Digit (TGLD) to exchange the active and held calls.

c. Dial the Disconnect Control Digit (DISD) to release the active call and reconnect the held call.

Programmable Control Digits

The Control Digits may be programmed in LD 15.

Three-party Service time out treatment

A timer is provided on a customer basis to activate an optional time out treatment. The optional time out treatment is to release the active party and connect to the held party if the controlling party of a Consultation call does not dial a Control Digit within the time specified during a Register Recall. The result is the same as if the controlling party had dialed the Control Digit assigned to the DISD function.

The optional time out treatment is selected by the user by responding to the Control Digit Time Out (CDTO) prompt in LD 15 with a value in the range of 2 to 14 seconds.

If the user selects either the default, 14 seconds, or enters 14 seconds, then the operation is as it was prior to the introduction of the Three-party Service time out treatment. That is, if a Control Digit is not entered within 14 seconds, Overflow Tone is provided for 14 seconds, after which the call returns to its previous state; the held party remains on hold and the consulted party is reconnected. If a Register Recall is performed while Overflow Tone is given the call returns to its previous state.

Six-party Conference Enhancement for analog (500/2500 type) telephones

Provides analog (500/2500 type) telephone users with the ability to Conference up to six parties.

If the MPO package is equipped, then Six-party Conference Enhancement is available to analog (500/2500 type) telephones with a combination of the TSA and existing C6A Classes of Service.

This capability is an extension of Three-party Service which allows the user to build a conference of up to six parties by consulting and selectively adding members through the use of Control Digits.

Ignore Switchhook Flash

Provides the ability, on a customer basis, to ignore a Switchhook Flash from analog (500/2500 type) telephones. This eliminates the confusion between a flash signal and a dial “1” signal on Dial Impulse analog (500/2500 type) telephones, especially when the Dial Impulse analog (500/2500 type) telephones have been assigned DTN Class of Service.

If the flash is to be ignored, analog (500/2500 type) telephones must have a Ground (EARTH) Button in order to use features which require a Register Recall.

Forced Register Recall

Provides an option, on a customer basis, to force analog (500/2500 type) telephone users assigned DTN Class of Service to issue a Register Recall before dialing a Control Digit.

If the system does not have the Forced Register Recall option activated, then a Switchhook Flash is interpreted as a dial “1”, default CNFD, causing that Control Digit assignment to be activated.

Manual return after enquiry (Manual Hold)

Provides an option (MHLD) to require analog (500/2500 type) telephone users to issue a Register Recall to return to a held party following a Consultation dialing time out.

At present, when an analog (500/2500 type) telephone places a party on hold by using a Register Recall, sets with DIP Class of Service receive 30 seconds, while Digitone (DTN) Class of Service sets receive 14 seconds, of Special Dial Tone followed by 14 seconds of Overflow Tone before the held party is reconnected. During this period, the held party is listening to silence, or RAN if equipped.

The Manual Return after Enquiry option (MHLD) controls the way the held party is reconnected to the controlling party. If MHLD = NO (the default), the controlling party is automatically reconnected to the held party after the overflow tone timeout. If MHLD = YES, the controlling party receives silence indefinitely after the overflow tone timeout until a second recall is performed to retrieve the held party. There is no automatic reconnection of the held party. The controlling party may manually return to the held party by performing a second Register Recall during Special Dial Tone, Overflow Tone or during the silence period.

Recovery of Misoperation during Call Transfer

The Recovery of Misoperation during a Call Transfer feature provides protection against having calls lost due to misoperation of the Call Transfer feature. Misoperation occurs whenever the user initiates an unexpected action that would normally cause a call to be lost.

If a station user tries to perform an illegal Call Transfer (for example, Call Transfer to a vacant number or Call Transfer to a busy extension), the station user receives the appropriate indication on the Consultation connection (for example, Overflow Tone and Busy Tone). However, since transfer in the ringing state is allowed, the user may still mis-operate and complete the Call Transfer operation immediately after dialing the desired number.

If a Meridian 1 proprietary telephone user attempts to complete a Call Transfer by pressing the Call Transfer key, the call is only transferred if the dialed party is in the ringing state or in the Consultation state with the controlling party. In other states the attempt to Call Transfer is ignored.

When an analog (500/2500 type) (500/2500) set initiates a supervised call transfer to a DN in any other state than ringing, the call transfer misoperation treatment is dependent upon the option chosen for AOCS (all other cases) in the customer data block (LD 15). If any set (either analog or digital) attempts a blind transfer in ringing state, the misoperation treatment is dependent upon the option chosen for Ring No Answer (RGNA) type of misoperation assigned in LD 15.

A number of options are available, where a call is transferred while the transferred station is ringing. For example if Attendant After Recall (AAR), or Disconnect After Recall (DAR) is selected, the transferred station will ring for an optional number of ring cycles (RCY2). On the expiration of this timer, the transferring set is rung back for an optional number of ring cycles (RCY1) with an optional recall ringing cadence. If the transferring station does not answer during the optional ringing cycles (RCY1), the transferred call will be forwarded to the attendant or Night Service DN (AAR) if external or disconnected by the DAR option if internal.

The Recovery options are specified for both Ring No Answer (RGNA) and All Other Cases (AOCS) cases in LD 15 when the MPO package is equipped. Separate treatment can be specified for external and internal calls.

Switchhook Contact Bounce

The situation occurs when an analog (500/2500 type) telephone goes on-hook. Switchhook contact bounce during disconnect may be interpreted by the system as a switchhook flash followed by an on-hook. When this occurs there is an unintended Call Transfer to the attendant or other type of misoperation.

In order to resolve this problem, with the MPO package equipped, the software is modified to delay recognition of any action for a minimum of 256 milliseconds following receiving a valid switchhook flash from analog (500/2500 type) telephones. During this delay, any signaling received from the parties involved is ignored.

Operating parameters

For enhanced functionality of the Multi-Party Operations, the following features should be equipped:

- Automatic Hold for Meridian 1 proprietary telephones
- Ground Button and Flash timers, and
- Recall of misoperation ringing cadence and Control and Special Dial Tones requires the Flexible Tones and Cadences (FTC) feature.

Feature interactions

Access to Paging trunks

Analog (500/2500 type) telephones with TSA Class of Service are restricted from initiating a Consultation connection while connected to a paging trunk.

Access to Recorded Dictation trunks

Analog (500/2500 type) telephones with TSA Class of Service are restricted from initiating a Consultation connection while connected to a dictation trunk.

Attendant Administration

Attendant Administration allows certain station classes of service to be altered. The operation of Attendant Administration is modified so that if an attendant tries to alter either Call Transfer Allowed (XFA) or Call Transfer Denied (XFD) Class of Service, then Three-party Service (TSA) Class of Service is disallowed. The TSA and XFA Classes of Service are mutually exclusive. When XFA is assigned, TSA will be disallowed if it was not configured. XFD is not mutually exclusive with TSA, but TSA will not be automatically assigned if the Class of Service is changed to XFD. TSA Class of Service cannot be assigned using Attendant Administration.

This feature can not be used to setup the Three-party Service TSA Class of Service.

Attendant Break-In

Break-In is not allowed to the party receiving the patience tone or the misoperation ringback.

Break-In with Secrecy

For Multi-Party Operation (MPO), the operation of features, such as going on-hook and releasing from a call, during the BKIS conference between the attendant and the desired party, takes precedence over MPO operations for those cases where the treatment differs from that defined by the customer.

All network nodes must have MPO software, with identical Multiple-party Operation (MPO) options. Otherwise, MPO options in the desired party's node have precedence.

Pertaining to MPO options, if the undesired party is not located on the same node as the desired party, the undesired party is considered as an external party on the desired party node.

Attendant Forward No Answer

Multi-Party Operations – Recovery of Misoperation During Call Transfer takes precedence over NFNA and NFNS for DID/DOD/CO calls.

When a DID/DOD/CO call is transferred from one station to another station on the same node, Ring Again No Answer has priority over NFNA and NFNS.

Attendant Recall

For analog (500/2500 type) telephones with TSA Class of Service, Attendant Recall is accomplished by performing a Register Recall during the two-party connection and dialing the Attendant DN.

Attendant Recall with Splitting

The Multi-Party Operations (MPO) feature introduces a new Class of Service, Three Parties Service Allowed (TSA), for analog (500/2500 type) telephones. It allows certain keys on these sets to be programmed for conference, toggle between sets, and disconnect. However, the toggle function will be disabled if a call is transferred to the attendant because of the Attendant Recall with Splitting feature.

Call Forward All Calls

A set which has activated Call Forward All Calls can still initiate calls and become the controlling party of a consultation connection. In this case, if the set mis-operates, then Multi-Party operations, while re-ringing the controlling party as a part of misoperation recovery, ignores the Call Forward All Calls indication present on the controlling party.

Call Forward No Answer

For Call Transfer with Ring No Answer (RGNA) if the user has selected an option other than Standard, the optional treatment has priority over the CFNA option selected in the LD 15. If the user has chosen the standard option for RGNA, the call will be treated as a normal CFNA call, and handled according to the options selected for CFNA in LD 15. Once the call is routed to a Night DN during recovery of misoperation and the Night DN does not answer, the call is treated according to the NFNA and FDN options chosen for the Night DN. The Night DN can use flexible CFNA DN in two levels. MPO misoperation does not change the operation of the DNFD timer if one has been configured in LD 15.

Call Pickup

Analog (500/2500 type) telephones with Call Pickup Allowed (PUA) and TSA Class of Service can pick up a call only if they are not involved in another call. After picking up a call, the user can form a Consultation connection and dial Programmable Control Digits as normal.

Call Pickup, Directed

Users of analog (500/2500 type) telephones involved in a Three-Party Service call cannot pick up another call by dialing the SPRE code.

Analog (500/2500 type) sets with TSA Class of Service, which are actively involved in Three-party Service, are not allowed to dial the Special Prefix (SPRE) code to pickup another call.

Call Transfer

Analog (500/2500 type) telephones with TSA Class of Service perform a supervised Call Transfer by going on-hook after establishing a conference. This differs from operation with XFA Class of service, where transfer can be achieved by going on-hook during Consultation connection. If an analog (500/2500 type) telephone with TSA Class of Service goes on-hook during consultation connection, it is treated as misoperation of All Other Cases and the recovery actions are done based on the CCDO and AOCS options selected in LD 15. If CDOC = NO, an analog (500/2500 type) telephone can achieve a transfer by going on-hook after establishing a conference.

During the Consultation connection, the non-controlling parties are restricted from using Call Transfer, Conference, and Three-party Service features.

Call Waiting

An analog (500/2500 type) telephone may be assigned both Call Waiting Allowed (CWA) and TSA Classes of Service. The user can establish a Consultation connection by answering Call Waiting during an active established call. If this is done, Control Digit features (Conference Digit (CNFD), Toggle Digit (TGLD), and Disconnect Digit (DISD)) are available. Note that Programmable Control Digit TGLD, rather than a switchhook flash, is used to toggle the calls. Operation with XFA Class of Service is unchanged.

The Three-party Service feature changes the operation of Call Waiting for all analog (500/2500 type) telephones as follows (regardless of whether the sets have TSA Class of Service. If an analog (500/2500 type) telephone user activates Waiting during an active call so as to establish a Consultation connection, and if the user goes on-hook during the Consultation connection, the operation is treated as an AOCS misoperation. The recovery of misoperation will take place even if the MPO package is not equipped. In this case, the controlling party will be re-rung by the held party regardless of the Consultation Connection Disconnect Option (CCDO) and the recovery of misoperation options.

If an analog (500/2500 type) telephone user attempts to set up a Consultation connection by dialing a busy DN and if the Call Waiting conditions are satisfied, the controlling party will hear ringback tone and the active party will hear Call Waiting tone. If the controlling party goes on-hook before the active party has answered, the held call is disconnected regardless of the MPO options and Call Waiting tone is removed from the active party.

Call Waiting Redirection

Recovery on Misoperation of Call Transfer – Call Transfer with Ring No Answer (RGNA)

With the Call Waiting Redirection feature enabled, if the Controlling Party goes on-hook to complete the call transfer before the Active Party answers the Call Waiting call, and before the CFNA timer applied Call Waiting Redirection feature times out, there is no change.

With the Call Waiting Redirection feature enabled, if the CFNA timer applied by the Call Waiting Redirection feature times out before the Call Transfer completes in the Ring No Answer (RGNA) state, CFNA treatment is given by the Call Waiting Redirection feature only if the RGNA option is defined to be Standard (that is, operation as it was prior to the introduction of the Multi-Party Operations feature).

For Call Transfer with Ring No Answer, if the user has selected an option other than Standard treatment, the RGNA option selected has priority over the CFNA option selected in the Customer Data Block. With the Call Waiting Redirection feature enabled, the non-Standard RGNA option will also be enforced. There are no interactions in the functioning of Multi-Party Operations for the Attendant After Recall, Disconnect After Recall, Attendant After Recall, Overflow, and Disconnect RGNA call treatment options.

As the transferred set tries to re-ring the transferring set, if the transferring set is busy, call redirection will again try Call Forward All Calls, Hunting, and Call Waiting in that order. Call Waiting Redirection will not apply CFNA treatment to the unanswered Call Waiting call as the non-Standard RGNA option selected has priority over the CFNA option selected in the Customer Data Block, and thus have priority over Call Waiting Redirection CFNA treatment.

Recovery on Misoperation of Call Transfer – Misoperation of Call Transfer for All Other Cases

This type of misoperation occurs when the transferring party attempts to complete the transfer in several other non-RGNA scenarios. There is no interaction with these Multi-Party Operations scenarios and the Call Waiting Redirection feature.

Camp-on

Camp-on to a controlling party DN which is involved in a Consultation connection is not permitted. However, Camp-on is allowed at non-controlling parties DN's which are involved in the Consultation connection.

Camp-On, Forced Override

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

China – Supervised Analog Lines

As in the cases with Call Transfer and Conference, the call type of the first active call determines whether battery reversal or hook flash supervision applies. Also, supervision signaling is not supported for the second call. A disconnect supervision signal is extended only when the last party disconnects.

Supervised Analog Lines

The call type of the first active call determines whether battery reversal or hook flash supervision applies. Also, supervision signaling is not supported for the second call. A disconnect supervision signal is extended only when the last party disconnects.

China – Toll Call Loss Plan

When a user toggles between one party and another, the Toll Loss Plan is inserted on the active call if it is a toll call. If the user toggles to a non-toll call, the Toll Loss Plan is removed.

Conference

Current Conference feature for analog (500/2500 type) telephones with C6A is not affected by conference with TSA Class of Service.

The Call Join feature allows a user of a system or digital telephone to conference in or transfer a third party to a party held on the user's telephone, without having to dial the third party. The user can then hang up.

The patience tone or the Misoperation ringback is not applied to a conference party.

Display of Calling Party Denied

When three parties are joined using the Call Join capabilities of the Multi Party Operations feature, display information is not provided on any of the conferee's sets. When setting up a conference call, by conferencing one set at a time, the display on the conferee's set is in accordance with the individual set's Class of Service. If one set leaves a three party conference, display information on the remaining sets is based on the individual Class of Service of each set.

End-to-End Signaling

The party receiving the patience tone or the Misoperation ringback is not able to use End-to-End Signaling.

Enhanced Music on Hold

Analog (500/2500 type) telephones with TSA Class of Service can receive music when put on hold during Three-party Service.

Enhanced Night Service

Enhanced Night Service allows a mis-operated call involving a Direct Inward Dial (DID) trunk to queue at the Night Service DN.

Group Hunt

As per the existing Multi-Party Operations (MPO) feature, recovery of misoperation of call transfer will not be applied to incoming calls which are transferred on ringing to a Pilot DN by transferring parties who are waiting in GPHT queues for service.

Last Number Redial

For analog (500/2500 type) telephones with TSA Class of Service, the first call of a Consultation connection is stored as the last number. Last Number Redial (LNR) is possible whenever Dial Tone or Special Dial Tone is given.

Night Service

If the system is in Night Service mode, mishandled calls which are routed to the attendant are rerouted to the appropriate Night Service DN. External trunk calls, other than DID, are queued till they are answered.

TIE trunk calls are not queued at the Night Service DN. If the Night Service DN is busy, TIE calls are disconnected.

Off-hook Alarm Security

Three-party Service (TSA) and Alarm Security Allowed (ASCA) Classes of Service are mutually exclusive. A set assigned TSA Class of Service cannot also be assigned ASCA Class of Service, and vice versa; a set assigned ASCA Class of Service cannot also be assigned TSA Class of Service.

Override, Enhanced

With Priority Override (POVR) equipped, there is a slight change in Multi-Party Operations functionality. When a consultation call is made without POVR equipped, and the telephone being called is busy, a recall returns to the party on hold without dialing a control digit. However, if POVR is equipped, a control digit must be dialed. Any control digit releases the busy call and returns to the call on hold.

Paging

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a paging trunk.

Recall to Same Attendant

Users of analog (500/2500 type) telephones can perform an attendant recall during a two-party connection by performing a switchhook flash and then dialing the attendant DN.

Recorded Telephone Dictation

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a dictation trunk.

Ring Again

When a TSA Class of Service analog (500/2500 type) telephone with a call on hold encounters Busy Tone, Ring Again is not possible.

Slow Answer Recall Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Slow Answer Recall for Transferred External Trunks

The Multiple Party Operation recall can only be applied in a standalone environment, and therefore does not interact with this feature.

Stored Number Redial

For analog (500/2500 type) telephones with TSA Class of Service, the current LNR number can be stored only after the Consultation connection is completely released. Save Number Redial (SNR) is possible whenever Dial Tone or Special Dial Tone is given.

Tone to Last Party

When the MPO package is equipped, Tone to Last Party is not provided.

Trunk to Trunk Connection

In a standalone environment, the RGNA prompt in the Customer Data Block will be used when an external trunk is transferred on ringing and the called party does not answer. In a network environment, the Recall Timers (RTIM) timer value in the Customer Data Block will be used for slow answer recall.

Feature packaging

The basic Multi-Party Operations features are packaged under Multi-Party Operations (MPO) package 141.

For enhanced functionality of the Multi-Party Operations feature, the Flexible Tones and Cadences (FTC) package 125 is required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the Multi-Party Operations parameters in the Customer data block.

2 LD 10 – Assign Three party Service (TSA) Class of Service to sets.

3 LD 56 – Configure Control Dial Tone and recall Tones and Cadences cadences for analog (500/2500 type) telephones and Meridian 1 proprietary telephones.

LD 15 – Configure the Multi-Party Operations parameters in the Customer data block.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	MPO	Multi-Party Operations.
...		
- FMOP	YES	Change Flexible misoperation Parameters.
- - RGNA	xxx yyy	Ring No Answer. Enter treatment for Call Transfer Ring No Answer cases, where: xxx is the treatment for internal parties, and yyy is the treatment for external parties.
	(STD) (STD)	– (default for internal and external parties) Standard treatment
	AAR AAR	– Attendant After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles route call to an attendant.
	ATN ATN	– Attendant: Route call to an attendant.
	DAR DAR	– Disconnect After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles disconnect call.
	DIS DIS	– Disconnect: disconnect call.
	OVF OVF	– Overflow tone: Call is given Overflow Tone.

- - AOCS	xxx yyy	All Other Cases. Enter treatment for Call Transfer cases other than Ring No Answer: xxx is the treatment for internal parties and yyy is the treatment for external parties.
	AAR AAR	– Attendant After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles route call to an attendant.
	ATN (ATN)	– (default treatment for external parties) Attendant: Route call to an attendant.
	DAR DAR	– Disconnect After Recall: Recall transferring (controller) set for RCY1 number of ring cycles. If transferring set does not answer within RCY1 ring cycles disconnect call.
	(DIS) DIS	– (default treatment for internal parties) Disconnect: disconnect call.
	OVF OVF	– Overflow tone: Call is given Overflow Tone.
	STD STD	— Standard treatment ***Note:*** If entered in response to AOCS in LD 15, responses will be printed as DIS ATN in LD 21.
- - RCY1	1-(6)-15	Ring Cycles 1. Number of ring cycles (default is 6) a transferring (controlling) station is rung before routing to an attendant or disconnect occurs.
- - RCY2	1-(4)-15	Ring Cycles 2. Number of ring cycles (default is 4) target (transferred to) station is rung before Ring No Answer treatment is applied. Does not apply to AOCS.
- - RALL	YES	Recall YES – Mandatory recall is required prior to dialing Control Digits.
- - CDTO	2-(14)	Control Digit Time Out. Range is 2 - 14 seconds and inputs must be a multiple of 2, (that is, 2, 4, 6, 8, 10, 12, or 14). 2 to 12 activates the optional time out treatment. 14 (default) activates the normal time out treatment.

- IFLS	(NO), YES	Ignore switchhook flash. NO – (default) Allows a switchhook flash, or dial "1", from an analog (500/2500 type) telephone to be interpreted as a Register Recall. YES – A switchhook flash, or dial "1", from an analog (500/2500 type) telephone will not be interpreted as a Register Recall. ***Note:*** If this option is selected, analog (500/2500 type) telephones should be equipped with a special Ground (Earth) Button.
- MHLD	(NO), YES	Manual Hold. NO – (default) Manual hold is not allowed. YES – Manual hold is allowed.
- PCDS	(NO), YES	Program Control Digits. YES – Allows user to alter default settings of Control Digits. NO – (default) Does not allow the alteration of the existing Control Digit settings. CCDO is the next prompt. Programming of control digits is not required. The default is NO. The defaults values for their respective functions are 1, 2 and 3. If YES then:
- - CNFD	0-(1)-9, *,#	Conference Digit. Prompted if response to PDCS was YES. Enter the Control Digit used to create, or add parties to, a conference. Default is 1.
- - TGLD	0-(2)-9, *,#	Toggle Digit. Prompted if response to PDCS was YES. Enter the Control Digit used to toggle, put active party on hold and connect to held party/parties. Default is 2.
- - DISD	0-(3)-9, *,#	Disconnect Digit. Prompted if response to PDCS was YES. Enter the Control Digit used to disconnect the active party and connect to the held party. Default digit is 3.

- CCDO	(NO), YES	Consultation Connection Disconnect Option. NO – (default) Alternative treatment is not applied to Consultation calls where one of the parties disconnects. YES – Alternative treatment is applied to Consultation calls where one of the parties disconnects.
- AFNO	(NO), YES	(Manual) Forced Camp-On Automatic.
- - ACNS		Attendant Clearing during Night Service. Prompted when the MPO package is equipped and MPOP and FMOP = YES.
	(NO) EXT ALL	No automatic treatment. External calls only. All calls.

LD 10 – Assign Three party Service (TSA) Class of Service to sets.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Type of telephone.
...		

CLS	TSA	Three-party Class of Service Allowed. TSA and ASCA are mutually exclusive (that is, if TSA is assigned then ASCA will not be allowed, and vice versa). TSA interacts with XFA in the following manner. If the set has XFA (Call Transfer Allowed) Class of Service and the administrator then assigns TSA (Three-party Service Allowed), XFA (Call Transfer Allowed) is automatically set to XFD (Call Transfer Denied) and Three-party Service is then allowed. Conversely if the set has TSA Class of Service assigned and the administrator then assigns XFA, Three-party Service is removed and Call Transfer is allowed. The last Class of Service entered overwrites the previously entered Class of Service of the same category (that is, if both XFA and TSA are entered in that order, TSA is the Class of Service that is accepted.)
...		

LD 56 – Configure Control Dial Tone and recall Tones and Cadences cadences for analog (500/2500 type) telephones and Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG NEW PRT	Change, add, or print.
TYPE	FTC	Flexible Tones and Cadences data block.
TABLE	0-31	FTC table number.
...		
RING	YES	Change the ringing feature definitions.
...		
- PCAD	xxx	Recall of misoperation ringing Cadence Enter Master Cadence (MCAD) table number that defines the ringing cadence for recall of misoperation for analog (500/2500 type) telephones and Meridian Modular sets. Default is value assigned to NCAD.

- PBCS		Recall of misoperation ringing tone and cadence for Meridian 1 proprietary telephones. Define recall of misoperation tone and cadence for SL-1 and M1000 sets.
TDSH	i bb cc tt	Tone and Digit Switch Hexadecimal code. Prompted if Tone and Digit Switches (TDS) are configured in LD 17. Defaults are values assigned to NBCS.
XTON XCAD	0-255 0-255	Extended tone code. Extended cadence code. Respond to the XTON prompt with a value from 0 to 255, for the NT8D17 TDS Tone code. Respond to the XCAD prompt with a value from 0 to 255, for the NT8D17 TDS cadence code for FCAD. Prompted if system configured with Extended Conference and Tone and Digit Switches (XCT) in LD 17. Defaults are the values assigned to NBCS.
...		
HCCT	YES	Hardware Controlled Cadences and Tones.
...		
- CDT		Control Dial Tone. Define tone and cadence for Control Dial Tone.

TDSH	i bb cc tt	Tone and Digit Switch Hexadecimal code. Prompted if Tone and Digit Switches (TDS) configured in LD 17. Defaults are values assigned to DIAL.
XTON XCAD	0-255 0-255	Extended tone code. Extended cadence code. Respond to the XTON prompt with a value from 0 to 255, for the NT8D17 TDS Tone code. Respond to the XCAD prompt with a value from 0 to 255, for the NT8D17 TDS cadence code for FCAD. Prompted if system configured with Extended Conference and Tone and Digit Switches (XCT) in LD 17. Defaults are the values assigned to DIAL.

Note: Refer to *Software Input/Output: Administration* (553-3001-311) for complete information regarding the administration of tones and cadences.

LD 81 – This overlay is modified to print the stations associated with Three-party Service Allowed (TSA) Class of Service if the MPO package is equipped.

LD 83 – This overlay is modified to include the TSA Class of Service, when sorting TN by Class of Service, if the MPO package is equipped.

Feature operation

Prior to describing the feature operation the following terms are defined to ensure there is no misunderstanding as to their meaning in terms of the Multi-Party Operations feature.

Active party – The party with which the controlling party has a Consultation connection.

Analog (500/2500 type) set – For the purpose of this document, this term is used to refer to standard analog (500/2500) sets.

Meridian 1 proprietary telephone – For the purpose of this document, this term is used to refer to standard SL-1 sets and to digital sets (M2000 series and M3000).

Bridged sets – The "Bridging" feature allows the same DN to appear on more than one single line telephone. Bridged sets share the same TN. Up to eight of these sets can be bridged, and a maximum of five of this group can be equipped with ringers. An incoming call rings all sets that have ringers connected and can be answered by any single line telephone user within the bridged group.

Controlling party – The "Controlling Party" is the party which optionally has the "Held Party" in the hold mode and the "Active Party" in the "Consultation Connection".

Consultation connection – When the controlling party and the active party are in conversation, they are said to be in "Consultation Connection".

Dial "1" – A pulse recognized as digit 1.

External party – Any CO, DID or TIE trunk (incoming or outgoing), connected to the system is considered an external party, regardless of the way the connection is established.

Flash Timer – The Flash Timer defines the flash period of a valid Switchhook Flash.

Held party – The Held party is the party put on hold (by the Controlling party).

Programmable Control Digit – A digit which is dialed by the controlling party, after the Consultation connection is established, to achieve certain functions of Three-party Service for an analog (500/2500 type) telephone.

Register Recall – A user request for service produced either by Switchhook Flash or by pressing the Ground Button or the Link button.

Switchhook Flash – An on/off-hook pulse which may be either a Register Recall signal or a Digit 1 depending on the conditions during which it occurs and on the flash timing.

Call Join

Call Join is available on any Meridian 1 proprietary telephone that is equipped with a Three-party (AO3) or Six-party (AO6) Conference key and at least one secondary DN or Call Waiting key.

The following describes the operation of Call Join:

- If the user presses the AO3 or AO6 key during an active call with party A on DNx (DNx is any DN key, including Call Waiting), party A is placed on hold and Special Dial Tone is returned as normal. The user can dial another DN and conference as normal or the user can conference a held party B on DNy (DNy is any DN key, excluding DNx) by continuing as follows:

  Note: M2317 and M3000 set soft keys may not display correctly.

- The user presses DNy during Special Dial Tone. This causes party B to be moved to the Conference key. DNy key is idled. The Conference key remains active and the user consults with party B.
- When the user has finished consulting with party B, the user presses the Conference key a second time. Party A, party B and the user form a conference (subject to normal restrictions) on DNx. The Conference key is idled. If the user disconnects during the conference, party A is transferred to party B, subject to normal restrictions.

The conference can be enlarged by operating the AO6 key either as described above to add a held party to the conference, or as normal to conference a dialed party.

Note 1: The DNx or secondary DN key can be any Meridian 1 proprietary telephone key capable of holding an independent Directory Number.

Note 2: If the Call Waiting is a Group Call, that call cannot be joined.

Analog (500/2500 type) telephone features

Multi-Party Operations introduces Three-party Service Allowed (TSA) Class of Service. Analog (500/2500 type) telephones can now be assigned TSA Class of Service and either C6D (Conference 6-party Denied) or C6A (Conference 6-party Allowed) Class of Service. Analog (500/2500 type) telephone operation is not changed for XFD or XFA Classes of Service.

Three-party Service permits the user to toggle, release or form a three-party conference through the use of Programmable Control Digits.

The combination of TSA and Conference 6-party (C6A) Classes of Service extend the operation of Three-party Service so as to permit the user to enlarge the three-party conference by consulting and selectively adding members through the use of Programmable Control Digits.

The following sections describe Three-party Service (TSA Class of Service).

Establishing a Consultation connection

If the user requests a Register Recall during any established two-party connection, excluding calls to Dictation or Paging trunks or to an attendant, the call is placed on hold and Special Dial Tone is returned. The user can dial a second party for Consultation.

If the controlling party goes on-hook before the second call is established (that is, when the transferred station is ringing, the call is treated as per Misoperation of Call Transfer).

When the second call is established, the user becomes the controlling party of the “Consultation” connection. The user can modify the connection through the use of a Programmable Control Digit.

Dialing a Control Digit from a Dial Impulse analog (500/2500 type) telephone with DIP or DTN Class of Service

After the consultation connection is established, the controlling party can dial a Programmable Control Digit. Here, if RALL = NO, both sets with DIP and DTN Class of Service dialing using dial impulses can dial the programmable control digits without performing the recall. However, for a dial impulse sets with DTN Class of Service the mode of dialing control digits depends upon how the set has setup the consultation call. If the set has used pulse dialing, then the control digits are recognized without recall. If the set has used touchtone dialing, Register Recall is mandatory.

If RALL = YES, a register recall must be performed prior to dialing a control digit, regardless of the set's Class of Service.

1 Dialing the Conference (CNFD) Control Digit produces a three-party conference between the user, held and active parties. During the Conference connection, all parties are restricted from using Call Transfer, Three-party Service and Three-party Conference features (unless the user has C6A Class of Service). If the user goes on-hook during the conference, the remaining parties stay connected as a normal two-party call, subject to normal restrictions.

2 Dialing the Toggle Control Digit (TGLD) exchanges active and held parties. During the Consultation connection, the controlling party is restricted from adding other parties to the call, and the non-controlling parties are restricted from using the Call Transfer, Conference and Three-party Service features.

3 Dialing the Disconnect Active Control Digit (DISD) releases the active party. The connection to the held party is automatically restored as a normal two-party connection. Either party can initiate another Consultation or Conference connection, subject to normal restrictions.

If the user dials any other digit, the connection to the active party is restored and the held party remains on hold.

Dialing a Control Digit from a Dual-tone Multifrequency analog (500/2500 type) telephone with DTN Class of Service

After the Consultation connection is established, the controlling party can dial a Control Digit. If the controlling party is a Dual-tone Multifrequency (DTMF) analog (500/2500 type) telephone with DTN Class of Service, a Register Recall must precede the Programmable Control Digit.

When the controlling party performs a Register Recall, the speechpath to the active party is removed. If no Digitone Receivers (DTRs) are available, no tone is given and the active party is reconnected. If a DTR is found, a new tone, Control Dial Tone, is given to the controlling party. The cadence, level and frequency of Control Dial Tone are flexible and defined on a per-customer basis.

During Control Dial Tone, the user can dial a Programmable Control Digit.

If a disconnect signal is received from the held party during Control Dial Tone, or if the user does not dial a Programmable Control Digit within 15 seconds, the DTR is removed and Overflow Tone is given for 14 seconds. During this time, the controlling party can restore the connection to the active party by performing a switchhook flash. At the end of Overflow Tone, the active party is reconnected and the held party (if still connected) remains on hold.

If the user performs a switchhook flash during Control Dial Tone, the connection with the active party is restored and the held party remains on hold.

Dialing a Control Digit from a Bridged Set

If Dial Impulse analog (500/2500 type) telephones and DTMF analog (500/2500 type) telephones are bridged and assigned DTN Class of Service, the operation depends on whether the Consultation connection was set up using Dial Impulse or DTMF.

If the Consultation connection was set up using Dial Impulse, only Dial Impulse analog (500/2500 type) telephone users can dial a Programmable Control Digit. If the Consultation connection was set up using DTMF, only DTMF analog (500/2500 type) telephone users can dial a Programmable Control Digit.

Any dial pulses or Register Recalls are recognized only if all other sets on the bridged line are on-hook. A Register Recall performed by using a Ground Button is also recognized.

Controlling party actions

The following table summarizes the affect on Consultation connections when controlling parties with XFA or TSA Class of Service perform the following actions:

Table 86
Control Digit results based on Class of Service

Controlling party action	System Response Class of Service	
	XFA	TSA
Dial CNFD	Conference	Conference
Dial TGLD	Conference	Toggle
Dial DISD	Conference	Release Active Party
On-hook	Transfer	Disconnect

Note 1: Dial Impulse analog (500/2500 type) telephones with DIP Class of Service are required to issue a Register Recall prior to dialing Control Digits if RALL = YES.

Note 2: If Control Dial Time Out is any value other than the default (14), then the time out results in the same action as if DISD had been dialed.

Note 3: If CCDO is YES, Call Transfer takes place when the controlling party goes on-hook during a consultation connection. This is similar to XFA operation.

Consultation Call Disconnect

Active Party Disconnects

If the disconnect during Consultation connection default (CCDO = NO) option is chosen after a Consultation connection has been established, then the active party disconnects if:

- The active party is internal to the circuit switched network or is external and a disconnect signal is received by the circuit switched network, the held party is reconnected for a normal two-party connection.
- The active party is external to the circuit switched network and a disconnect signal is not received by the circuit switched network, then the controlling party is able to release the disconnected trunk by dialing the Disconnect Active (DISD) Programmable Control Digit. The connection to the remaining party then becomes a normal two-party connection.

If the disconnect during Consultation connection alternative treatment (CCDO = YES) option is chosen, and the active party goes on-hook during an enquiry call, then the controlling party is given Overflow Tone. On tone time out or Register Recall, the held party is reconnected. If the controlling party goes on-hook during Overflow Tone, the call is treated as in Controlling Party Disconnects.

Held Party Disconnects

If the disconnect during Consultation connection option chosen is the default, after a Consultation connection has been established, the held party disconnects if:

a. The held party is internal to the circuit switched network or is external and a disconnect signal is received by the circuit switched network, then the connection with the active party becomes a normal two-party connection.

b. The held party is external to the circuit switched network and a disconnect signal is not received by the circuit switched network, due to the fact that the remaining connection is effectively a two-party connection, the trunk to which the departed party was connected is still on hold. The controlling party can release the disconnected trunk by dialing the Toggle (TGLD) Programmable Control Digit (to hold the active party and activate the connection to the disconnect trunk) and then dialing the Disconnect Active (DISD) Programmable Control Digit (to release the trunk). The connection to the remaining party becomes a normal two-party connection. If the controlling party goes on-hook with the disconnected trunk on hold, the set is rung back.

In the case of Procedure a (above), when a Dial Impulse analog (500/2500 type) telephone with DIP Class of Service user dials a Programmable Control Digit during the active call, Special Dial Tone is returned, indicating that the held party has disconnected. Similarly, when a DTMF analog (500/2500 type) telephone with DTN Class of Service user performs a switchhook flash during the active call (expecting to receive Control Dial Tone), Special Dial Tone is returned, indicating that the held party has disconnected.

During Special Dial Tone, the controlling party has the option of dialing a DN to set up another Consultation connection, or of resuming the normal two-party connection. The latter is achieved by performing a Register Recall with a duration greater than 150 milliseconds and less than the maximum flash time (a short Register Recall would be mistaken for a digit “1”). A dial “1” from a Dial Impulse analog (500/2500 type) telephone with DIP Class of Service cannot be used to simulate the flash, as the digit “1” may be the first digit of a DN. The user can do a valid switchhook flash during the middle of dialing a DN and be returned back to he held party. The only restriction is that the switchhook flash must be unambiguous (that is, the duration of the switchhook flash is greater than the digit “1” duration).

If a 2500 set recalls during a consultation connection and the held party has disconnected with the held party being an internal party or the system has received a disconnect signal, special dial tone is returned instead of control dial tone. This is similar to CCDO = NO.

If RALL = YES, the above operation also applies to 500 sets.

With RALL = NO and a 500 set (dial impulse) dials a control digit other than DISD, the set is given overflow tone indicating that the held party has disconnected. If the 500 set dials the DISD control digit, the active party is disconnected and the control party gets overflow tone.

Controlling Party Disconnects

If the disconnect during Consultation connection option chosen is default, then if the controlling party goes on-hook during the Consultation connection, it is considered as a misoperation of All Other Cases type (AOCS) and the active party is released.

DIS, ATN, AAR, DAR, and OVF options are available for both internal and external parties. If the held party is internal to the circuit switched network, the held party is optionally (DIS) released also. If the held party is external, the controlling set is optionally (AAR) rung back immediately. The external party does not receive Ringback Tone while the controlling set is being rung.

If the controlling party answers, the external party is connected for a normal two-party connection. If the controlling party does not answer within the optional ring cycles (RCY1) for any call (regardless of whether the set has FND or FNA Class of Service), the controlling station is idled while the external party receives Ringback Tone and is optionally routed to the attendant and appears on the CFNA Incoming Call Indicator. Other options are also available.

If the disconnect during Consultation connection option chosen is to give the alternative treatment, then if the controlling party goes on-hook during conversation with the active party, the call is transferred (as current operation with XFA Class of Service on the station).

Six-party Conference

The combination of C6A and TSA Classes of Service, provides an enhancement to the Six-party Conference feature where the user can perform a Register Recall during the Conference connection, dial a consulted party and then dial a Programmable Control Digit to toggle, release, or add the consulted party to the conference. The following describes the sequence of events required of an analog (500/2500 type) telephone with C6A and TSA Classes of Service to set up a multi-party conference:

a. During a normal two-party connection with party A, the user performs a Register Recall and dials party B. The user becomes the controlling party of the Consultation connection.

b. The user dials the Conference (CNFD) Programmable Control Digit to form a three-party conference. The Consultation connection becomes a Conference connection.

c. The user performs a Register Recall during the Conference connection. The conference is placed on hold (the other parties in the conference remain connected) and Special Dial Tone is returned. The normal timing and misoperation procedures apply while setting up the Consultation call. The user dials party C. When the Consultation call is established, the user becomes the controlling party of the new Consultation connection.

d. The user can dial a Programmable Control Digit which is interpreted as follows:

– Dialing CNFD causes the consulted party to be added to the conference, as shown in Procedure b. The Consultation connection becomes a Conference connection.

– Dialing TGLD causes the consulted party to be placed on hold and the conference to be reconnected. The user can toggle between the conference and the consulted party in this manner.

– Dialing DISD causes the consulted party to be disconnected. The Conference connection is restored.

The user can repeat Steps 3 and 4 to add parties to the conference. If the user goes on-hook during the Consultation connection, the consulted party is released and the conference stays connected, subject to normal restrictions. Six-party Conference Enhancement for analog (500/2500 type) telephones follows the same operation as the existing Six-party Conference feature with respect to misoperation, access and connection restrictions.

Recovery of misoperation during Call Transfer

Call Transfer with Ring No Answer (RGNA)

RGNA is applicable only when the user transfers a call while the active party is still in ringing state. All other types of misoperation are handled as AOCS misoperations.

Call treatment is then determined by the response to the RGNA prompt in LD 15. The following is a list of the responses to the RGNA prompt and the resulting treatment:

a. STD (Standard) – The operation as it was prior to the introduction of the MPO feature.

b. ATN (Attendant) – The transferred party is routed to the attendant if the target (transferred to) station, after having rung for an optional number of ring cycles (RCY2), has not answered the call. The call is rerouted to the attendant as a Call Forward No Answer (CFNA) and is presented on the FNA Incoming Call Indicator (ICI), the call is then treated as a regular CFNA call to the attendant.

c. If the transferred call was a Consultation connection the transferred party is disconnected and the held party is routed to an attendant and presented as a Recall on the RLL ICI.

d. DAR (Disconnect After Recall) – The target station rings for an optional number of ring cycles (RCY2). If the call is not answered during this time, the transferred party recalls the transferring (controlling) station. The transferring station rings for an optional number of ring cycles (RCY1), with recall ringing cadence. If the transferring station does not answer during this time, the transferred party is disconnected.

e. If the transferred call was a Consultation connection then the held party is retrieved and treated as defined by its type (internal or external) and the treatment selected. If the treatment selected is ATN or AAR the held party is routed to an attendant and presented as a Recall on the RLL ICI. If the treatment selected is DAR or DIS, the party is disconnected.

f. If the transferring station became busy before recall, the transferred party is disconnected immediately.

g. AAR (Attendant After Recall) – This option is similar to the DAR option, except that after the optional number of ringing cycles (RCY1) the transferred party is routed to an attendant as a Call Forward No Answer (CFNA) recall and is presented on the CFN ICI.

h. If the transferred call was a Consultation connection then the held party is retrieved and treated as defined by its type (internal or external) and the treatment selected. If the treatment selected is ATN or AAR the held party is routed to an attendant and presented as a Recall on the RLL ICI. If the treatment selected is DAR or DIS, the party is disconnected.

i. If the transferring station became busy before recall, the transferred party is routed to attendant immediately.

j. OVF (Overflow) – Overflow Tone is given to the transferred party after the optional number of ring cycles (RCY2).

k. If the transferred call was a Consultation connection, the transferred party is disconnected and the held party is given Overflow Tone.

l. DIS (Disconnect) – The transferred party is disconnected after the optional number of ring cycles (RCY2).

m. If the transferred call was a Consultation connection the transferred party and held party are disconnected.

Note: The ring cycles are counted from the time the transfer has been completed (analog (500/2500 type) telephone has gone on-hook or Meridian 1 proprietary telephone has pressed the TRN key for the second time).

This feature applies to both external and internal calls, transferred by station users to another station. The feature does not apply to calls transferred to the attendant, or extended by the attendant.

Misoperation during Call Transfer – All Other Cases (AOCS)

This section describes misoperation during Call Transfer for All Other Cases (AOCS) and their default options. Similar options as for Ring No Answer (RGNA) are available for AOCS. The only difference being that the ringing cycle (RCY2) is not valid for AOCS.

Call Transfer to a Busy Station

If an analog (500/2500 type) telephone user tries to transfer a call to a busy station, Busy Tone is returned during the Consultation connection. If the user then goes on-hook to complete the Transfer operation and if the held party is an external trunk, the external trunk is routed automatically to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall call to the attendant.

If the held party is an internal call, it is disconnected.

Call Transfer to Intercept Treatment

While using the Call Transfer feature, the analog (500/2500 type) telephone user may be intercepted while dialing the third party due to any of the following illegal dialing situations:

a. Dialing a vacant number.

b. Dialing a number of a terminal in the maintenance busy or RPE failure state.

c. Access denied.

d. Code Restriction or Toll Restriction.

e. Invalid, restricted, or blocked Network Automatic Route Selection (NARS) or Basic Automatic Route Selection (BARS) calls.

In any of the above cases, while involved in the Consultation connection (according to the selected customer option) the user is:

- given Overflow Tone

- given an intercept recorded announcement or
- routed to the attendant

If the user goes on-hook while connected to Overflow Tone or recorded announcement, and if the held party is an external trunk, the external trunk is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

If the MPO package is equipped and the user waits until time out occurs while connected to Overflow Tone or a recorded announcement, the held party is reconnected to the station user, and the call is treated as a regular two-party call again.

If the MPO package is not equipped, and the user waits until time out occurs while connected to Overflow Tone or a recorded announcement, both the internal and external calls are disconnected.

Unsuccessful Transfer Connection

While transferring an external trunk to another destination from an analog (500/2500 type) telephone, if network blocking prevents the completion of the Call Transfer or if the controlling party dials the access code of a busy trunk route, the controlling party receives Overflow Tone during the Consultation connection. If the analog (500/2500 type) telephone user goes on-hook in spite of the blocking indication, the external trunk is routed to the attendant as an Intercept Recall. At this point, the call is treated as a regular Intercept Recall to the attendant.

Call Transfer on Partial Dialing

If an analog (500/2500 type) telephone user dials an incomplete number as a third party and attempts to complete the Transfer operation by going on-hook, and if the held party is an external trunk, the external trunk is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

Disconnect Situations during Consultation

If the analog (500/2500 type) telephone user (the controlling party) disconnects while in the Consultation state, the call is transferred as normal. However, if the new connection is not possible (for example, due to trunk-to-trunk connection restrictions), and if the held party is external, then this external party is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

Also, if the analog (500/2500 type) telephone user (the controlling party) disconnects while connected to Dial Tone, and if the held party is external, then this external party is routed to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

If one of the other parties in the call disconnects, the following occurs:

- If the held party disconnects, the controlling party receives no indication until the hook switch is flashed to establish a conference. At that time Dial Tone is returned instead of all three parties creating a conference. The call is treated as a normal two-party call from the time the held party disconnects.
- While an external party is in the Consultation hold state, if the party being consulted disconnects followed by the controlling party disconnect, then the held party is routed automatically to the attendant as an Intercept Recall. The call is then treated as a regular Intercept Recall to the attendant.

Misoperation during Control Dial Tone

The treatment given depends upon the type of active party. If the active party is internal, the internal option is also applied to the held party (for example, if for internal calls AOCS is DIS ATTN, the held call even though external will also be disconnected). The misoperation option selected in this case is solely dependent upon the type of active call (internal or external), and the related misoperation option. This option is consistently applied to the held, as well as the active party.

With the Consultation Connection Disconnect Option (CCDO) in LD 15 not selected, if an analog (500/2500 type) telephone user (the controlling party) disconnects while receiving Control Dial Tone in the Consultation state, internal held parties are disconnected while external parties are routed to the attendant as Intercept Recalls. The external calls are then treated as a regular Intercept Recalls to the attendant.

With CCDO selected, if an analog (500/2500 type) telephone user (the controlling party) disconnects while receiving Control Dial Tone in the Consultation state the held parties are given treatment as defined by the responses to the All Other Cases (AOCS) prompt in LD 15.

Misoperation Treatment Options

A number of misoperation treatment options are made available both for internal and external calls. These treatment options are available for Ring No Answer (RGNA) and for All Other Cases (AOCS). The following are the cases for AOCS:

- Call Transfer to Intercept Treatment for:
  - Call Transfer to busy station
  - Dialing a vacant number
  - Terminal is in maintenance busy
  - RPE failure state
  - Access denial
  - Code or Toll restricted set
  - Network blocking
  - Invalid, restricted and blocked Network Automatic Route Selection (NARS)/Basic Automatic Route Selection (BARS) calls
  - Partial dialing
  - Trunk-to-trunk connection restrictions
  - Inter-tenant blocking
  - During reception of announcements, and
  - During reception of tones (Control, Special),

- Call Transfer while Dial Tone is being heard
- Call Transfer before completing dialing
- Call Transfer during outpulsing of digits on a trunk, and
- Controlling party goes on-hook during Consultation connection (CCDO = NO).

Recall of misoperation Ringing Cadence Option

When a transferring set is rung back after Call Transfer misoperation, then Recall of misoperation ringing cadence is optionally given to this set. Two optional Recall of misoperation cadences, one for analog (500/2500 type) telephone and Meridian Modular sets (PCAD) and another for SL-1 and M1000 series sets (PBCS), are optionally selectable (in LD 56). The default Recall of misoperation ringing cadence is the current Ringing Tone or cadence.

Multi-Party Operations Enhancements

Contents

This section contains information on the following topics:

Feature description 1059
Operating parameters 1060
Feature interactions 1060
Feature packaging 1061
Feature implementation 1061
Feature operation 1062

Feature description

The following enhancements pertain to the Three-party Service capability of Multi-Party Operations (MPO). Refer to the Multi-Party Operations feature description contained in this document for a description of Three-party Service.

Patience Tone

The controlling party may modify a Consultation connection by performing a Register Recall and then entering a Control Digit. During the call modification, this enhancement provides a “Patience” tone to the party on Consultation hold, rather than silence.

Ringback to external parties after misoperation

If the controlling party goes on-hook as a misoperation, the controlling set is rerung immediately. This enhancement allows the external party to receive ringback tone while the controlling party is rerung after misoperation.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Break-in

Attendant Break-in is not allowed to a connection in which a party is receiving Patience Tone or recall of misoperation ringback.

Call Transfer

A party receiving Patience Tone or recall of misoperation ringback is not able to Call Transfer.

Call Waiting

An analog (500/2500 type) telephone cannot have Call Waiting during Patience Tone.

Camp-on
Periodic Camp-on

While Camp-on and Periodic Camp-on are allowed on a party receiving Patience Tone, Camp-on tone and Periodic Camp-on tone are not applied to the party during Patience tone. However, Camp-on tone and Periodic Camp-on tone are applied when the speechpath has been reestablished.

Conference

Patience tone or recall of misoperation ringback are not applied to a conference party.

End-to-end Signaling

A party receiving Patience Tone or recall of misoperation ringback is not able to invoke End-to-end Signaling.

Multi-Party Operations

Usually the party on Consultation hold receives silence; with this improvement it will receive Patience Tone.

After a misoperation, when the controlling party is rerung and the far end receives silence, this improvement will provide ringback tone.

Feature packaging

These enhancements are packaged as part of the Supplementary Features (SUPP) package 131.

French Type Approval (FRTA) package 197 is also required to provide ringback tone to the held party while the controlling party is being rerung.

Feature implementation

LD 56 – Define Patience Tone and cadences.

Prompt	Response	Description
REQ	NEW CHG	Add new data, or change existing data.
TYPE	FTC	Flexible Tones and Cadences
...		
HCCT	YES	Hardware Controlled Cadences and Tones
- TLPT	(0)-30	Tone to Last Party Timer in seconds
- PATI		Patience tone Define Patience Tone and cadence

Prompt	Response	Description
TDSH	i bb cc tt	Tone and Digit Switch Hexadecimal code. Prompted if Tone and Digit Switch (TDS) is configured in LD 17. Default is (0000) no tone.
XTON XCAD	(0)-255 (0)-255	Extended Tone code. Extended Cadence code. Respond to the XTON prompt with a value from 0 to 255, for the NT8D17 TDS tone code. Default is 0. Respond to the XCAD prompt with a value from 0 to 255, for the NT8D17 TDS cadence code for FCAD. Default is 0. Prompted if system configured with Extended Conference and Tone and Digit Switches (XCT) in LD 17. Default is no tone.
...		

Note: Refer to *Software Input/Output: Administration* (553-3001-311) for complete information regarding the administration of tones and cadences.

Feature operation

Patience Tone to Consultation Held party during Control Dial Tone

To initiate Three-party Service analog (500/2500 type) telephones must perform a Register Recall, (that is, Switchhook Flash).

When the controlling party has established a Consultation connection, there is a call on hold and the Consultation connection is active. The controlling party can modify the connection through the use of a Control Digit.

To modify the call the controlling party performs a Register Recall, if the response to RALL in LD 15 is YES, to receive Control Dial Tone for 15 seconds. If no digit is dialed within 15 seconds the controlling set then receives Overflow Tone. If no digit is dialed, the controlling set is eventually put in lockout state.

The current operation is when a controlling party performs the Register Recall, the speechpath to the consulted party is removed, and the consulted party receives silence.

This enhancement allows a Patience Tone to be given to the consulted party instead on silence while the speechpath is removed.

Ringback sent when the controlling party is rerung after a misoperation

Current operation is when a controlling party goes on-hook and the on-hook constitutes a misoperation, the initial held call or the held consultation party may re-ring the controlling set immediately if the appropriate option (either AAR or DAR) is active. The external party does not receive ringback tone while the controlling set is being rung.

This enhancement allows a ringback tone to be provided to the external party when the controlling set is being rerung.

Multiple Appearance Directory Number Redirection Prime

Contents

This section contains information on the following topics:

Feature description 1065
Operating parameters 1066
Feature interactions 1067
Feature packaging 1073
Feature implementation 1074
Feature operation 1080

Feature description

Multiple Appearance Directory Number (DN) Redirection Prime (MARP) standardizes call redirection on Multiple Appearance DNs (MADNs) by using a service changeable Multiple Appearance DN Redirection Prime Terminal Number (MARP TN).

Each defined Single or Multiple Appearance DN has only one associated MARP TN. When a call redirection feature activated against a DN needs Terminal Number (TN) specific information, the MARP TN is used to determine feature operation. Call redirection always refers to the MARP TN.

MARP provides consistent operation for the following call redirection features:

- Call Forward All Calls

- Call Forward Busy
- Call Forward No Answer, and
- Hunting.

Operating parameters

All systems support a maximum of 30 appearances of the same DN.

Short Hunt takes precedence over MARP TN directions.

MARP is activated in LD 17. If MARP is not active, refer to specific call redirection modules in this document for call redirection details. MARP prompts and messages appear even if MARP is not active. MARP TNs can still be added, assigned, and changed.

The MARP TN is defined in LD 10 or LD 11. When activated, only the MARP TN is used to determine call redirection.

If MARP is not activated, the overlays listed have this message printed: "MARP NOT ACTIVATED." The message appears only once, when the overlay is loaded. When MARP is active, no message appears. The overlays affected are: LDs 10, 11, 20, 22, 25, 80, 81, 82, and 83.

When MARP is activated in Service Change (MARP = YES), calls are immediately directed according to the MARP TN. There is no need to SYSLOAD.

Every Single or Multiple Appearance DN has a MARP TN. MARP TNs are also defined for Data DNs, optional incoming two-way Hot Line DNs, and ringing and nonringing Private Line DNs. Automatic Call Distribution (ACD) DNs are not assigned MARP TNs.

New systems are installed with MARP activated. MARP TNs are assigned to all Single and Multiple Appearance DNs. Call redirection follows the MARP TN assignments.

MARP TNs assigned at Service Change

Each DN must have an associated MARP TN. After a Service Change or a telephone relocation, the system assigns a MARP TN to the DN in the following situations:

- The MARP TN containing the DN is removed.
- The DN appearance on its MARP TN is changed to another DN.
- The DN appearance on its MARP TN is no longer the redirection prime.

The "TN list" refers to the list of TNs that appears when you print the DN block in LD 20 or LD 22 (TYPE = DNB). To determine the order in which your TNs appear, print out the DN block.

When assigning MARP TNs during Service Change, the system conducts a search beginning at the top of the TN list for the first appearance of the DN as the Prime DN. The MARP TN is assigned based on the following:

- The first TN found with a primary appearance of the DN is assigned as the MARP TN.
- If no primary appearance of the DN is found, the first TN encountered with a secondary appearance of the DN is assigned as the MARP TN.

Feature interactions

Attendant Administration

MARP TNs cannot be added, moved, or deleted with Attendant Administration. The DN information that displays on the console includes the MARP designation if applicable.

Attendant administration activities, like changing key assignments or DN appearance, may change MARP TN assignments. If so, CSC102 appears on the teletype (TTY) indicating a new default MARP TN, as follows:

CSC102 DN nnnn NEW MARP l s c u

where
nnnn = the DN associated with the MARP TN
l s c u = the new MARP TN assigned to DN nnnn

Attendant and Network-Wide Remote Call Forward (RCFW)

The RCFW feature operation applies only to one prime DN of a Multiple Appearance DN. If multiple stations are configured with the same prime DN, the set-based network RCFW feature operation is the same as the standalone RCFW feature operation.

If multiple stations are assigned the same prime DN and station control password (SCPW), the RCFW operation applies to the station to which the MARP TN is assigned. If none of the stations is configured as the MARP TN for that prime DN, the Remote Call Forward Activate and Deactivate Flexible Feature Codes (FFCs) apply to all stations matching the DN and SCPW. Remote Call Forward Verify applies to the station according to MADN call presentation priority, placing the station with the last service change at the end of the list.

The attendant-based RCFW operation applies to the station with the MARP TN of the DN entered.

Attendant Break-In

The attendant may get a busy tone if all the telephones with the required DN are busy. Attendant Break-In permits the attendant to break in to the connection with the least restricted TN. Where more than one TN exists that meets this criterion, Break-In chooses the one at the bottom of the DN block.

Automatic Set Relocation
Modular Telephone Relocation

When Automatic Set Relocation is used to move a telephone, the telephone's MARP designations are maintained. During the relocation, a temporary MARP TN is assigned. The original MARP TN is restored when the telephone relocates.

- When a telephone leaves the system due to set relocation, the following Customer Service Change (CSC) message appears:

  CSC010 x y

  where
  x = old TN (l s c u) for the telephone
  y = ID code entered

- The following Service Change (SCH) message appears for any MARP TN reassignment:

  SCH5524 DN nnnn NEW MARP l s c u

  where
  nnnn = the DN associated with the MARP TN
  l s c u = the new default MARP for DN nnnn

- The History File can be configured to store these messages until a printout is requested.

- When a telephone reenters the system, the following message appears:

  CSC011 x y

  where
  x = old TN (l s c u) for the telephone
  y = new TN (l s c u) for the telephone

- The following message appears again for each changed TN:

  SCH5524 DN nnnn NEW MARP l s c u

  where
  nnnn = the DN associated with the MARP TN
  l s c u = the new MARP TN assigned to DN nnnn

Automatic Call Distribution

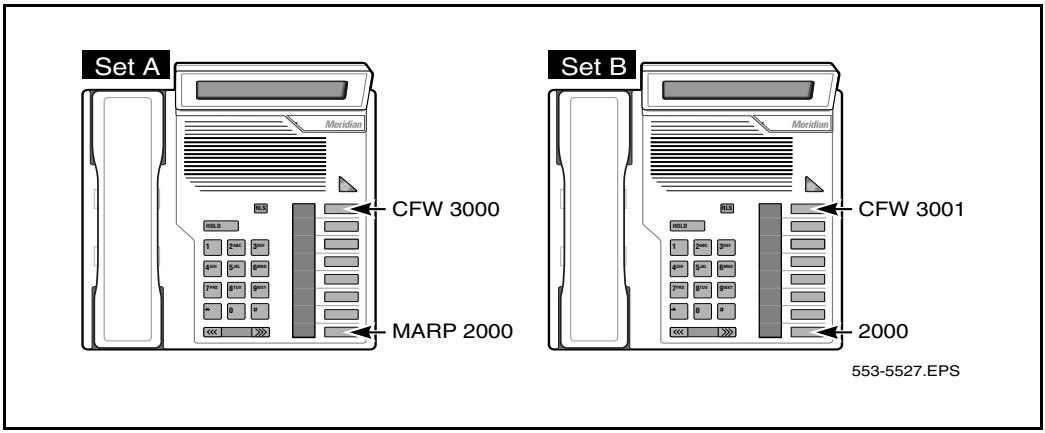
Automatic Call Distribution (ACD) DNs are not assigned MARP TNs. Agent Individual DNs (IDNs) are assigned MARP TNs.

Call Forward All Calls

If CFW is active for a DN, incoming calls are forwarded if a TN is found that has CFW enabled and is a single appearance or a prime multiple appearance of that DN (according to existing operation). The MARP TN is always checked first to meet these criteria. When the requirements are met, the system uses the information associated with the MARP TN to redirect the call.

If the MARP TN is not a prime appearance but does have CFW enabled, a search is made for a telephone with a prime appearance of that DN with CFW enabled. When a TN is found, the call is redirected according to the MARP TN's parameters. If the MARP TN is not a prime appearance and does not have CFW enabled, the system searches for a prime appearance with CFW enabled. The incoming call is forwarded according to the other telephone's instructions (not the MARP TN's), as shown in Figure 70.

Figure 70
CFW and MARP

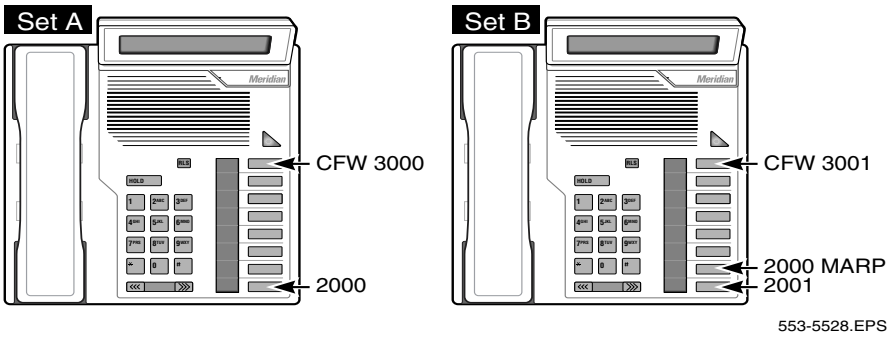


CFW DN on Telephone A is DN 3000. CFW DN on Telephone B is DN 3001.

- If only Telephone A has CFW active, calls to DN 2000 are forwarded to DN 3000.
- If only Telephone B has CFW active, calls to DN 2000 are forwarded to DN 3001.
- If both Telephone A and B have CFW enabled, calls to DN 2000 are forwarded to DN 3000 because Telephone A is the MARP TN.

At times, even though the MARP TN is actually a secondary DN appearance, it can control where a call is redirected. Due to potential confusion, it is recommended that a secondary appearance not be defined as the MARP TN when a prime appearance is available. Refer to Figure 71.

Figure 71
MARP control



CFW DN on Telephone A is DN 3000. CFW DN on Telephone B is DN 3001.

- If both Telephone A and Telephone B have CFW active, all calls to DN 2000 go to DN 3001 because Telephone B is the MARP TN.
- If only Telephone A has CFW active, all calls to DN 2000 go to DN 3000.
- If only Telephone B has CFW active, no calls to DN 2000 are forwarded.

  If all DN appearances are secondary, no calls are forwarded.

Call Forward No Answer

The MARP TN always controls the call redirection for Call Forward No Answer.

- If a DN is assigned as a Prime DN on a telephone and as a secondary DN on one or more telephones, the DN list is still organized as described in the preceding paragraphs. If only one prime appearance of a DN exists, however, call redirection parameters are derived from the TN of the prime appearance telephone, even though it may not be at the end of the list. A prime appearance is always the first TN used when the system looks for call redirection instructions.

- If a DN appears on analog (500/2500 type) telephones, and Meridian 1 proprietary telephones, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the list. Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the beginning of the list. A service change to a Meridian 1 proprietary telephone moves its TN to the end of the list.
- A SYSLOAD restructures the list back to numerical TN order with analog (500/2500 type) telephones at the top and Meridian 1 proprietary telephones at the bottom. Call redirection parameters continue to be derived as described in the preceding paragraphs.

Call Redirection by Time of Day (CRTOD)

When CRTOD and Multiple Appearance DN Redirection Prime (MARP) are activated, Call Forward or Hunt are dependent on the time of day and follows the MARP feature for Call Forward No Answer or Hunt treatment.

Call Waiting Redirection

If the Multiple Appearance Directory Number Redirection Prime (MARP) feature is activated, the Call Forward No Answer (CFNA) treatment given by Call Waiting Redirection for an unanswered Call Waiting call follows the MARP feature for CFNA treatment of calls to an idle DN.

Electronic Lock Network Wide/Electronic Lock on Private LIne s

The same locked or unlocked state applies to all Terminal Numbers with the same primary DN and the same SCPW. Terminal Numbers with the same DN, but not having the same SCPW, cannot be locked or unlocked.

Hunting

The MARP TN always controls the call redirection for Hunting. Short Hunting takes precedence over Hunting and MARP. The MARP TN is referred to until Short Hunting is encountered. Short Hunting is in control until it expires. When short hunting expires, the MARP TN for the first DN in the Short Hunt sequence takes control.

Network Intercom

If more than one set is allocated the same prime DN, the Hot Type I call will terminate on the set designated as the Multiple Appearance Redirection Prime (MARP). If the MARP DN is not the prime DN on the set, or if the set designated as the MARP DN is not a Meridian 1 proprietary telephone, the first Meridian 1 proprietary telephone with the prime DN will be used. If none of these conditions are met, the call will terminate as a non-Hot Line call and the calling party will be notified on the display.

Hot Type D calls can have voice termination only on a MARP Terminal Number (TN), or if there is no MARP TN, then on the first TN in the TN list. A No Answer Indication for Hot Type D can only be left on the MARP TN, or if there is no MARP TN, then on the first TN in the TN list.

Phantom Terminal Numbers (TNs)

Multiple appearance and MARP cannot be enabled on a phantom TN.

User Selectable Call Redirection

When a Multiple Appearance DN is rung, the determination of the number of ringing cycles for CFNA depends on the value of the MARP prompt in LD 17. If the value is “YES,” the number of ringing cycles is determined by the Ringing Cycle Option (RCO) number of the DN that is classified as a MARP TN. If the DN is a Multiple Appearance DN (MADN), the RCO values in the other TN blocks for that DN are ignored.

If the MARP value is “NO,” the RCO is taken from the first TN in the DN block with a primary appearance of the DN. If there is none, the last TN in the DN block is used.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Activate or deactivate MARP.

2 LD 10 – Add an analog (500/2500 type) telephone with a Single Appearance DN.

3 LD 10 – Add an analog (500/2500 type) telephone with a Multiple Appearance DN.

4 LD 10 – Change an analog (500/2500 type) telephone with a Multiple Appearance DN.

5 LD 11 – Add a telephone with a Single Appearance DN.

6 LD 11 – Add a telephone with a Multiple Appearance DN.

7 LD 11 – Change a telephone with a Multiple Appearance DN.

8 LD 10 – Remove a MARP TN.

9 LD 11 – Remove a MARP TN.

10 LD 20 or LD 22 – Print MARP information.

If MARP is not activated, the overlays listed have this message printed: "MARP NOT ACTIVATED." The message appears only once, at the very beginning of the overlay. When MARP is active, no message appears. The overlays are: LDs 10, 11, 20, 22, 25, 80, 81, 82, and 83.

When changing or adding a new Single Appearance DN to the system, the MARP TN is automatically assigned. The system indicates this TN is the MARP for the new DN with a MARP message.

When adding or changing a Multiple Appearance DN, the system indicates which TN is the current MARP TN. You can reassign the MARP TN if required.

SCH5524 appears at the end of the Service Change session, when the MARP TN has been changed.

LD 17 – Activate or deactivate MARP.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. Gate opener.
PARM	YES	Change system parameters.
- MARP	YES NO	Activate or deactivate MARP. There is no default. <CR> retains the previous system data.

LD 10 – Add an analog (500/2500 type) telephone with a Single Appearance DN.

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	500	500/2500 telephone.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DN	xxx...x	Directory Number.
- MARP		MARP prints on the next line indicating this TN is the MARP for DN xxxx.

LD 10 – Add an analog (500/2500 type) telephone with a Multiple Appearance DN.

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	500	500/2500 telephone.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

DN	xxx...x	Directory Number.
- MARP ON TN	l s c u c u	*MARP ON TN l s c u* prints on the next line indicating TN l s c u (c u for Small Systems and Succession 1000 systems) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.

LD 10 – Change an analog (500/2500 type) telephone with a Multiple Appearance DN.

Prompt	**Response**	**Description**
REQ:	CHG	Modify existing data.
TYPE:	500	500/2500 telephone.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DN	xxx...x	Directory Number.
- MARP ON TN	l s c u c u	This message indicates the current MARP is TN l s c u (c u for Small Systems and Succession 1000 systems).
- MARP	(NO) YES	(Do not) set the MARP to this TN.

LD 11 – Add a telephone with a Single Appearance DN.

Prompt	**Response**	**Description**
REQ:	NEW	Add new data to the system.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

KEY	xx aaa yyyy	xx is the key number aaa is the DN type: MCN (multi-call nonring) MCR (multi-call ring) SCN (single-call nonring), or SCR (single-call ring). yyyy is the DN.
- MARP		*MARP* prints on the next line indicating this TN is the MARP for DN yyyy.
KEY		Reprompts until <CR> is entered.

LD 11 – Add a telephone with a Multiple Appearance DN.

Prompt	**Response**	**Description**
REQ:	NEW	Add new data to the system.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx aaa yyyy	xx is the key number. aaa is the DN type: MCN (multi-call nonring) MCR (multi-call ring) SCN (single-call nonring), or SCR (single-call ring). yyyy is an existing DN.
- MARP ON TN	l s c u c u	*MARP ON TN l s c u* prints on the next line indicating TN l s c u (c u for Small Systems and Succession 1000 systems) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.
KEY		Reprompts until <CR> is entered.

LD 11 – Change a telephone with a Multiple Appearance DN.

Prompt	Response	Description
REQ:	CHG	Modify existing data
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx aaa yyyy	xx is the key number. aaa is the DN type: MCN (multi-call nonring) MCR (multi-call ring) SCN (single-call nonring), or SCR (single-call ring). yyyy is the DN.
- MARP ON TN	l s c u c u	*MARP ON TN l s c u* prints on the next line indicating TN l s c u (c u for Small Systems and Succession 1000 systems) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to the working TN.
KEY		Reprompts until <CR> is entered.

LD 10 – Remove a MARP TN.

Prompt	Response	Description
REQ:	OUT	Remove data from the system.
TYPE:	aaaa	Telephone type, where: aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 11 – Remove a MARP TN.

Prompt	Response	Description
REQ:	OUT	Remove data from the system.
TYPE:	aaaa	Telephone type, where: aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 20 or LD 22 – Print MARP information.

Prompt	Response	Description
REQ	PRT	Print information.
TYPE	TNB	Terminal Number data block.
	(DNB SL1)	(Can also print out DN data block or telephone type.)

The printout will look like the following.

- For the DN data block:

  DN 2000
  TYPE SL1
  TN 018 0 02 00 KEY 00 MARP DES NO DES NO DATE
  TN 018 0 02 01 KEY 01 DES NO DES NO DATE

- For a telephone data block:

  DES NO DES
  TN 001 0 0 00
  TYPE SL1
  KEY 00 MCR 2000 MARP
  01 MRK

Feature operation

No specific operating procedures are required to use this feature.

Multiple Appearance Directory Number

Contents

This section contains information on the following topics:

Feature description 1081
Operating parameters 1083
Feature interactions 1083
Feature packaging 1092
Feature implementation 1093
Feature operation 1093

Feature description

DNs can appear on more than one multiline telephone, and can be shared between those telephones and single-line telephones. Up to 30 appearances of the same DN are allowed on Large Systems only. Four multiple-appearance options are provided, as follows:

- Multiple Call Arrangement with Ringing (MCR)
- Multiple Call Arrangement without Ringing (MCN)
- Single Call Arrangement with Ringing (SCR), and
- Single Call Arrangement without Ringing (SCN).

The customer can specify which of the four options applies to each appearance of the DN.

Multiple Appearance Directory Numbers (MADNs) are not restricted to telephones connected to the same loop. Telephones with MADNs can be assigned to different loops if the Loop Removal enhancement is allowed in LD 17 under the prompt MLDN.

A Multiple Appearance, Multiple Call Arrangement is available between Meridian 1 proprietary telephones only. It allows as many calls to be in progress as there are appearances of the DN. Selection of the ring option allows the DN to be rung whenever an incoming call is directed to the idle DN.

Selection of the no ring option causes the DN appearance not to ring when an incoming call is directed to the DN. Indication of an incoming call is limited to a flashing lamp associated with the DN.

Multiple Appearance, Single Call Arrangement DNs allow a single call to be active on the DN, irrespective of its number of appearances. Multiple Appearance, Single Call Arrangement is available to all telephones.

Selection of the ring option allows ringing to accompany lamp flashing when a call is directed to a DN. Privacy is inherent in active calls, except in a mixed arrangement – analog (500/2500 type) telephones and Meridian 1 proprietary telephones with an appearance of the same DN.

Call redirection parameters such as Hunt and Call Forward No Answer are derived from the TN data block (LD 20 TNB) of the prime appearance of the called DN. If there is more than one prime appearance, the parameters are selected from the last TN in the DN block for the DN (LD 22 DNB).

If more than one prime appearance of an MADN exists, the information noted in the following list must be considered prior to configuring call redirection parameters for MADNs.

- The DNB organizes MADN information in numerical TN order. The TN with the highest numerical value (000-0-06-03) is placed at the beginning of the DN list. The list then continues in descending order with the lowest numerical TN (000-0-03-01) at the end of the list.
- If a telephone undergoes Service Change, the TN of the telephone is moved to the beginning of the DN list regardless of the numerical value of the TN. This telephone remains at the beginning of the list until another telephone undergoes Service Change or a SYSLOAD is performed. A SYSLOAD restores the DN list to numerical TN order.

- If a DN is assigned as a prime DN on one telephone, and as a secondary DN on one or more telephones, the DN list is still organized as described in the preceding text. However, if only one prime appearance of a DN exists, call redirection parameters are derived from the TN of the prime appearance telephone, even though it may not be at the end of the list. A prime appearance is always the first TN used when the system looks for call redirection instructions.
- If a DN appears on analog (500/2500 type), and Meridian 1 proprietary telephones simultaneously, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the DN list, and Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves the TN of that telephone to the beginning of the list. A service change to a Meridian 1 proprietary telephone moves the TN of the telephone to the end of the list. A SYSLOAD restores the list to numerical TN order, with analog (500/2500 type) telephones at the top of the list and Meridian 1 proprietary telephones at the bottom of the list. Call Redirection parameters continue to be derived as described in the preceding text.

It is not necessary to change any data to register service change activity. To put a telephone at the end of the list, simply call up the service change data and default through the data.

Operating parameters

Multiple Appearance, Multiple Call Arrangement is limited to Meridian 1 proprietary telephones. If telephones are mixed, only Multiple Appearance, Single Call Arrangement is allowed.

For Multiple Appearance, Single Call Arrangement, the no ring option is limited to Meridian 1 proprietary telephones.

Feature interactions

Automatic Redial

An ARDL call from a Single Call Ringing (SCR) or Single Call Non Ringing (SCN) is only redialed when all sets that have the same DN are free.

An ARDL call from a Multiple Call Ringing (MCR) or Multiple Call Non Ringing (MCN) is only redialed when the originating key is free.

Automatic Wake Up

All Multiple Appearance DNs are rung, including both primary and secondary DNs. Programming the wake up request using the Wake Up key applies only to telephones with the primary DN on key 0, and the Wake Up indicator operates as described only on the telephone that is currently programming the wake up request.

In addition, if two or more Multiple Appearance Primary DN telephones program a wake up request at the same time, the last telephone to finish overrides. All telephones with the same primary DN get the same request time of the last telephone to program a request. If the last telephone cancels the request, all requests are canceled.

When the wake up programming sequence is finished, all Wake Up indicators on Multiple Appearance Prime DNs are updated unless a telephone is in the middle of Wake Up programming.

If the AWU Recall option is chosen, the recall is presented to any idle Attendant Console in the same Console Presentation Group (CPG) equipped with the AWU key.

Automatic Wake FFC Delimiter

For Multiple Appearance Directory Numbers, wake up information is stored, deleted and queried from a DN's first primary appearance terminal number.

Call Detail Recording on Redirected Incoming Calls

If the DN of the set forwarding the call is a Multiple Appearance DN, the Terminal Number of the set will be printed out in the AUX ID field (that is, line two of the Call Detail Recording record).

Call Forward by Call Type
Call Forward No Answer, Second Level

Call redirection parameters like Call Forward No Answer are derived from the TN data block of the prime appearance of the called MADN. If there is more than one prime appearance, the parameters are selected from the last TN in the DN block.

If more than one prime appearance of a MADN exists, the following information must be considered prior to configuring call redirection parameters for MADNs.

The DN Block organizes MADN information in numerical TN order. The TN with the highest numerical value (000-0-06-03) is placed at the beginning of the list. The list then continues in descending order with the lowest numerical TN (000-0-03-01) at the end of the list. Service change activity affects the organization of the DN list as described in the following paragraphs.

- If a telephone undergoes Service Change, its TN is moved to the beginning of the DN list, irrespective of the numerical value. This telephone remains at the beginning of the list until another service change or a SYSLOAD.
- If a DN appears on analog (500/2500 type) telephones and Meridian 1 proprietary telephones, the analog (500/2500 type) telephones are listed in numerical TN order at the top of the list. Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A Service Change to an analog (500/2500 type) telephone moves its TN to the beginning of the list. A Service Change to a Meridian 1 proprietary telephone moves its TN to the end of the list.
- A SYSLOAD restructures the list back to numerical TN order, with analog (500/2500 type) telephones at the top and Meridian 1 proprietary telephones at the bottom. Call Redirection parameters continue to be derived as described in the preceding paragraphs.

Call Forward, Remote (Attendant and Network Wide)

The Call Forward, Remote (RCFW) feature only applies to the primary appearances of Multiple Appearance DNs, and it is recommended that only one appearance of a Multiple Appearance DN be configured as the prime DN.

For the case of multiple stations with the same prime DN and SCPW, the RCFW operation will apply to the station that has the Multiple Appearance Redirection Prime (MARP) assigned to it.

If none of the stations having the DN and SCPW assigned are configured as the MARP TN for that DN, the RCFA and RCFD will apply to all stations matching the DN and SCPW.

The attendant-based RCFW feature will only apply remote call forward operation to the prime DN with MARP status. If the DN is not the prime DN or does not have MARP status, overflow tone will be received by the user.

Calling Party Name Display Denied

For a ringing call to a Multiple Appearance DN, the name on the calling set display can be suppressed by configuring any of the Terminal Numbers with NAMD Class of Service. The digit display on the calling set cannot be suppressed – the called digits are displayed even though the Class of Service on any of the Terminal Numbers is DIGD. The called set display is subject to the Class of Service of the calling party. For an established call to a Multiple Appearance DN, the calling set display is subject to the Class of Service configured for the answering set. The answering set display only is subject to the Class of Service of the calling party – the displays of the other sets in the Multiple-appearance group are blank.

Call Waiting Redirection

The Call Waiting Redirection feature applies to unanswered Call Waiting calls which apply to single appearance DNs and primary appearance DNs of MADNs.

China – Attendant Monitor

If Attendant Monitor is attempted on a Multiple Appearance DN, the Multiple Appearance Redirection Prime (MARP) TN becomes the desired party.

Controlled Class of Service

Controlled Class of Service (CCOS) restriction levels are activated or canceled on controlled telephones through their Prime Directory Number (PDN). When the PDN of a Meridian 1 proprietary telephone is made CCOS active, all DNs on that telephone are also restricted. If the DN is a PDN on other telephones, those telephones are also restricted (if they have CCSA Class of Service).

Controlled Class of Service, Enhanced

All Controlled Class of Service (CCOS) restriction levels are activated and canceled from the Prime Directory Number (PDN) for CCOS controlling telephones. The PDN for an SL-1 telephone is made CCOS active, and all DNs for that telephone are restricted as well. If that DN is a PDN on other telephones, they are also restricted (if they have CCSA Class of Service).

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

If the attendant tries to extend a call to a DN which appears on more than one set, this DN can either be:

- Multiple-Call Arrangement with Ringing (MCR): when a call terminates on this DN, all idle stations on which the DN appears are rung. The call is established only with the station which has answered first. All others are idle.
- Multiple-Call Arrangement with No Ringing (MCN): the only difference between MCN and MCR is that the called stations are not rung (only their DN keys flash).
- Single-Call Arrangement with Ringing (SCR): when a call terminates on this DN, all idle stations on which the DN appears are rung. The call is established only with the station which has answered first. All others are busy.
- Single-Call Arrangement with No Ringing (SCN): the only difference between SCN and SCR is that the called stations are not rung (only their DN keys flash).

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Since the ANI category is defined on a per set basis, two stations with the same Multiple Appearance Directory Number (MADN) can be assigned different ANI categories.

Directory Number Expansion

The DN can have up to seven digits if the Directory Number Expansion package is equipped.

If Loop Restriction Removal is allowed, telephones with MADNs can be moved across loops using Automatic Set Relocation (LD 25), the Meridian 1 proprietary telephones data block (LD 11), the analog (500/2500 type) telephone data block (LD 10), or Attendant Administration.

Display Calling Party Denied

When a Multiple Appearance DN is ringing, the display of the calling telephone does not show the caller's name if at least one of the TNs has Named Denied (NAMD) Class of Service. The dialed DN is displayed even if one of the TNs has DN Denied (DDGD) Class of Service. The display of the called telephone shows the DN and the caller's name according to the Class of Service of the calling DN.

When a Multiple Appearance DN is answered, the display of the calling telephone shows the DN and caller's name and DN according to the Class of Service of the answering TN. The display of the answering telephone remains the same, while the displays of the other telephones are blanked.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The same locked or unlocked state applies to all Terminal Numbers with the same primary DN and the same SCPW. Terminal Numbers with the same DN, but not having the same SCPW, cannot be locked or unlocked.

Group Call

The maximum number of DNs that can be added as members of a Group Call is 20. Each Multiple Appearance, MCR/MCN DN reduces the number of telephone sets that can be added to a Group Call. For example, if two telephones have the same MCR appearance of a DN, the number of telephones in the Group Call becomes 19. That is, each appearance of a DN counts as one member, up to a maximum of 20, of the Group Call.

> ***Note:*** Multiple Appearance, SCR/SCN DNs count as one member of a Group Call, irrespective of its number of DN appearances.

Group Hunt

While Multiple Appearance DNs (MADN) single call arrangements are treated the same as Single Appearance DNs (SADN), MADN multiple call arrangements must be avoided in a group hunt list.

With MADN multiple call arrangement, the idle or busy status of the MADN is determined by the terminal number (TN) data block of the prime appearance of the called DN. If there is more than one prime appearance of the called DN, the idle or busy status is then selected from the last TN in the DN block for the MADN (DNB prompt in LD 22). This means that there may be idle appearances of the MADN, while the hunt cycle regards them as busy and attempts to terminate on the next idle member of the group hunt list. If a MADN multiple call arrangement has to be used, a supervisor set must be assigned to the hunt group. This supervisor set must be given the only prime appearance of the MADN. Any other appearance must have the MADN programmed as a secondary DN (any DN key other than 0). In this way, the supervisor set controls the status of the MADN and thus the group hunt treatment. If the supervisor set is busy, the hunt does not terminate on the MADN.

Hunting

Hunting can be controlled by the MADN Redirection Prime (MARP) Terminal Number (TN). If the MARP system option is disabled, Hunting proceeds as if MARP did not exist.

If all the telephones in the Multiple Appearance Directory Number (MADN) group are Meridian 1 proprietary telephones, ringing telephones are placed at the top of the DN list, and non-ringing telephones are placed at the bottom.

If a Multiple Appearance Directory Number appears in a group with several telephone types, the telephone type affects the position of the TN in the list. The analog (500/2500 type) telephones are listed at the top, and Meridian 1 proprietary telephones are listed in numerical TN order at the bottom of the list. A service change to an analog (500/2500 type) telephone moves its TN to the top of the list. A service change to a Meridian 1 proprietary telephone moves it to the bottom of the list. Call redirection follows the TN order from top to bottom.

The MARP TN is always checked to determine if and how the call is to be redirected by Hunting, regardless of where the MARP TN resides in the TN list of the DN block. No searching of the TN list of the DN block is needed. Hunting will follow the hunt chain based on the originally dialed DN. The actual functioning and requirements for Hunting are not changed by the MARP feature. The basic change introduced by the MARP feature is to always have a designated TN, the MARP TN, as the TN supplying the call redirection parameters.

If the MARP TN does not have Hunting control enabled, no Hunting is attempted. Other features for redirecting calls to busy DNs may be attempted based on the MARP TN.

A Short Hunting sequence begins when the MARP TN of a busy DN can perform Short Hunting. When a Short Hunt begins, it completes on that telephone before going to the Hunt DN. The precedence of Short Hunting over normal Hunting is maintained. Once a Short Hunting sequence is started on a digital TN, all the DNs in the Short Hunt sequence on that TN are attempted before redirecting the call to the TN's Hunt DN. Thus, a Hunt Chain connects Short Hunting sequences through Hunt DNs only.

Last Number Redial

A last number dialed on a Directory Number (DN) with multiple appearances is stored only against the telephone from which the number was originally dialed.

Loop Restriction

If Loop Restriction removal is not allowed, telephones with MADNs can be moved by using the Automatic Set Relocation feature (LD 25), or the Attendant Administration feature.

Meridian 911

The DN keys for multiple appearance sets can be defined as an SCR (single call ringing) key or as an MCR (multiple call ringing) key. For those DNs (keys on MADN sets) that are SCR, only one call may be answered at a time. That is to say that once a call taker has answered a call, future calls to that DN will receive busy tone until the call taker on that DN has disconnected.

For DNs that are MCR, calls will only be given busy tone once every call taker is busy answering a call. If one call taker is answering a call and there are other call takers available, a new call to that DN will cause the sets of the available call takers to ring. Any available call taker can then answer the new call.

Message Registration

For Multiple Appearance Directory Number (MADN), the system selects the appropriate meter for the DN based on following this procedure:

- It accesses the meter of the most recently configured telephone having a Prime DN (PDN) appearance and Message Registration Allowed (MRA) Class of Service.

If no Terminal Number (TN) in the DN block has MRA Class of Service, the customer meter is charged. For the Message Registration Key (MRK), the system provides overflow and sets the MRK lamp to flash. For the Background Terminal (BGD), it prints a NO DATA FOUND message.

Privacy

If a Multiple Appearance, Single Call Arrangement (SCR) or Single Call Arrangement without Ringing (SCN) DN is shared by Meridian 1 proprietary telephones only, Privacy is in effect. No one can enter a call unless the call is first placed on Hold, or unless Privacy Release is activated to allow another appearance to enter the call. If this configuration is shared between these telephones and single-line telephones, Privacy is not in effect for any appearance of the DN. Anyone sharing the DN can enter the call at any time.

Privacy Override

Since the Privacy feature is not active in this mode, telephones with a Privacy Override Denied Class of Service can bridge into an active call.

Privacy Release

Privacy Release has no effect on Multiple Appearance, Multiple Call Arrangement with Ringing (MCR), or Multiple Call Arrangement without Ringing (MCN) calls.

Remote Call Forward

With a Multiple Appearance Directory Number (DN) and both sets having a Station Control Password (SCPW), Remote Call Forward may not operate as intended (that is, if Call Forward has been activated using the Remote Call Forward feature, Call Forward remains activated when an attempt to deactivate it is made from the set on which it is active).

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Since the ANI category is defined on a per set basis for Three Wire Analog Trunks, two stations with the same multiple Appearance DN can be assigned different ANI categories.

Voice Call

If a Voice Call DN is added to a second telephone, the DN becomes a Multiple Appearance DN (MADN). Voice Call does not support MADN.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Assign a Multiple Appearance Directory Number key.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx MCN yyy...y	Add a multiple-call non-ringing DN key, where: xx = key number, and yyy...y = DN.
	xx MCR yyy...y	Add a multiple-call ringing DN key, where: xx = key number, and yyy...y = DN.
	xx SCN yyy...y	Add a single call non-ringing DN key, where: xx = key number, and yyy...y = DN.
	xx SCR yyy...y	Add a single call ringing DN key, where: xx = key number, and yyy...y = DN.

Feature operation

No specific operating procedures are required to use this feature.

Multiple Console Operation

Contents

This section contains information on the following topics:

Feature description 1095
Operating parameters 1097
Feature interactions 1097
Feature packaging 1097
Feature implementation 1097
Feature operation 1099

Feature description

The system permits each customer to have up to 63 Attendant Consoles. Incoming calls are routed in a circular fashion to the first idle attendant. If all consoles are busy, calls are held in the attendant queue and are presented to the first idle attendant. Each console is identified by a customer-defined, two-digit Attendant Console number (01 to 63).

The assignment of Incoming Call Indicators (ICIs) and Trunk Group Busy (TGB) key/lamp pairs is identical for all Attendant Consoles in the customer group, except when Console Presentation Group Level Services, a multi-tenant feature, is configured. The flexible features key/lamp strip can be assigned on a per console basis.

The features that can be assigned to the flexible features strip include the following:

- Attendant Administration

- Autodial
- Automatic Wake Up
- Barge-In
- Busy Verify
- Call Park
- Calling Party Number
- Charge Account
- Controlled Class of Service, Enhanced
- Display Calls Waiting
- Display Date
- Display/Change Date
- Display Destination
- Display Source
- Display Time
- Display/Change Time
- Do Not Disturb (Individual)
- Do Not Disturb (Group)
- End-to-End Signaling
- Malicious Call Trace
- Message Cancellation
- Message Indication
- Paging
- Routing Control
- Speed Call Controller
- System Speed Call Controller, and
- Stored Number Redial.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Departmental Listed Directory Number

Departmental Listed Directory Number (DLDN) supports the assignment of 63 consoles per DLDN.

Multi-Tenant Services

Up to 63 consoles can be defined in a single Console Presentation Group (CPG).

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 12 – Select Attendant Console number.

2 LD 15 – Select Supervisory Console.

3 LD 93 – Configure Multi-Tenant Service.

LD 12 – Select Attendant Console number.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT	Attendant Data Block.
...		
ANUM	1-63	Attendant Number.

LD 15 – Select Supervisory Console.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	ATT	Attendant Console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
SPVC	(0)-63	Supervisory Console.

LD 93 – Configure Multi-Tenant Service.

Prompt	Response	Description
REQ	NEW CHG	Change.
TYPE	a...a	Type of data block (a...a = ACG, CPG, CPGP, RACC, RACG, RCPG, TACC, TACG, TCPG, TENS or TGEN).
CUST	xx	Customer number, as defined in LD 15
CPG	1-63	Console Presentation Group number.
...		

AGNO	0-63	Attendant Console Group Number.
...		
ANUM	1-63 1-63	Add Attendant Console Numbers.
...		
NAGN	0-63	Night Attendant Console Group Number.

Feature operation

No specific operating procedures are required to use this feature.

Multiple Customer Operation

Contents

This section contains information on the following topics:

Feature description 1101
Operating parameters 1101
Feature interactions 1102
Feature packaging 1102
Feature implementation 1102
Feature operation 1102

Feature description

The system can serve up to 32 (customer numbers 0-31) individual customers from the same machine. The system software supports 100 customer groups (numbered 0-99). Customers have their own features, restrictions, numbering plans, trunks, and special services. They are granted access to the system as if they are the sole user.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

System hardware, like Serial Data Interface (SDI), Digitone Receiver (DTR), Tone and Digit Switch (TDS), and Conference, is shared among all the customers on the machine.

The Speed Call list parameter (8191) applies to the machine, not the customer. It is shared among all customers on the system.

Feature packaging

Multiple Customer Operation (CUST) package 2 has no feature package dependencies.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Multi-Site Mobility Networking

Contents

This section contains information on the following topics:

Feature description 1103
Operating parameters 1104
Feature interactions 1104
Feature packaging 1105
Feature implementation 1105
Feature operation 1110

Feature description

Multi-Site Mobility Networking (MSMN) allows a Companion DECT handset user to make and receive calls at any MCDN node. When the handset user visits a MCDN node the MSMN feature automatically:

- detects the visiting handset when it is on
- forwards calls to the visiting handset from the users home node

The call forward dial tone gives an indication when MSMN activation was not successful. The user can turn the handset off and on again to re-activate the MSMN feature.

The MSMN feature requires concentrated DMCs. The DMCs must be 8D to support concentration. A non-concentrated system has each handset configured to a DMC TN. A non-concentrated DMC has 32 handset TNs assigned to 32 time slots and is non-blocking. A concentrated system has each handset configured to a Virtual TN (VTN) on phantom loops. Concentration allows up to 510 handsets to share the DMCs 32 time slots and is a blocking system.

Separate DECT systems on a Meridian circuit switched network can be either concentrated or non-concentrated.

Operating parameters

The MSMN feature can not support a mix of concentrated DMCs and non-concentrated DMCs within the same Companion DECT system. All DMCs must have at least one handset configured.

Feature interactions

Call forward from a MADN handset

A MADN handset at a remote node can activate call forward at the home node. When the handset shares a DN with another set(s), the CFW lamp lights on the shared DN set(s). If the handset is not the MARP, the shared DN MARP set can cancel call forward. If the handset is the MARP, the handset overrides any call forward which is setup from other shared DN set(s).

Card audit

Card audit does not work with VTNs.

Network Message Service

The MSMN feature does not change the handling of unanswered network calls. The Meridian Mail or Call Pilot network mail service does not change with multiple DNs configured against a single mailbox. The visiting DN receives the message waiting indication at the visited site.

Feature packaging

This feature requires the following packages:

- Multi-Site Mobility Networking (MSMIN) package 370
- Meridian 1 CT2 Mobility Option (MCMO) package 240
- Phantom loop (PHTN) package 254
- Meridian Companion MC32 (MC32) package 350
- Flexible Feature Code (FFC) package 139

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Add a new DCS handset.

2 LD 10 – Copy DCS handsets on DMC.

3 LD 10 – Remove DCS handsets.

4 LD 10 – Convert handset type 500 to DCS.

5 LD 20 – Print actual DMC TN and virtual DMC TN list.

6 LD 81 – Print DCS features.

The sequence of actions required to set-up visitor handsets are as follows:

1 Configure a phantom superloop using LD 97, if required.

2 Create the new DCS sets in LD 10.

3 Configure the RCFW data in LD 57 and LD 15 for handsets assigned as a visitor.

4 Use the DECT manager to configure sets on the DMC.

5 Pre-subscribe the visiting handset one time at the MCDN node.

Note: Subscription includes both overlay configuration and DECT Manager configuration. For DECT Manager configuration, see the Meridian Companion DECT NTP.

LD 10 – Add a new DCS handset.

Prompt	Response	Description
REQ:	NEW NEW 1-255 CHG ECHG	NEW = Add a Digital Cordless Set NEW X = The generation of new DCS units stop when the maximum Index of 509 is reached on a single DMC or VTNs on the system run out or WRLS ISM limits reached. All new DCS must be on the same DMC. CHG = Allows the DCS configuration to change to another DMC. All new DCS must be on the same DMC. ECHG = This command can change either the VSIT response or the HMDN response.
TYPE:	DCS	Digital Cordless Set. Differentiates between analog sets and non-concentrated digital Companion DECT handsets. If TYPE=DCS, the system allocates the next available VTN, and WRLS defaults to YES and WTYP defaults to DECT. If package #350 is included, MWUN defaults to 32. CLS defaults to ERCA, allowing the Enhanced RCFW feature.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	(4D)	Card Density. Only valid value for IPE is 4D. Normal input is <CR>.
WRLS	YES	WiReLess analog Set - entry defaults to YES with no user input - value cannot be CHG'ed.
WTYP	DECT	Wireless TYPe - entry defaults to DECT with no user input - value cannot be CHG'ed.

MWUN	32	Maximum number of Wireless UNits - entry defaults to 32 with no user input - value cannot be CHG'ed. Note - if MWUN = 32, CDEN automatically changes to 8D, and prints as an 8D unit.
DMC	 l s c c	Location of the actual DMC. Assigns a TN to a DECT Mobility Card. For Large Systems For Small Systems and Succession 1000 systems
INDX	0 .. 509	DMC index to map the Virtual TN to a DMC TN. Starting index on DMC, each unit increments to the next available unit.
VSIT	(NO)/YES	ViSITing DECT set. Determines the difference between a local handset and a visiting handset. VSIT available if the MSMN Package is unrestricted. YES = visiting DECT set. NO = local DECT set.
HMDN	X...X	HoMe Directory Number. Sets the DN as a valid MCDN network DN. NMDN available if VSIT=YES.

LD 10 – Copy DCS handsets on DMC.

Prompt	**Response**	**Description**
REQ:	CPY 1-32	CPY n = The generation of new units stop when the maximum index of 509 is reached on a single DMC or VTNs on the system run out or WRLS ISM limits reached. All DCS must be on the same DMC.
DMC	 l s c c	Location of the actual DMC to copy For Large Systems For Small Systems and Succession 1000 systems

LD 10 – Remove DCS handsets.

Prompt	Response	Description
REQ:	OUT 1-255	OUT X = Removing units stops when the maximum index of 509 is reached on a single DMC. All new DCS must be on the same DMC.
DMC		Location of the actual DMC to out on an IPE shelf or Small System cabinet.
	l s c	For Large Systems
	c	For Small Systems and Succession 1000 systems

LD 10 – Convert handset type 500 to DCS.

Prompt	Response	Description
REQ:	CDCS	Convert Digital Cordless Set - convert from a non-concentrated to a concentrated system after software upgrade. The conversion routine converts the 500 units to DCS units and moves them from the actual TN to a virtual TN.

Note: To convert from concentrated to non-concentrated, OUT all DCS units and re-subscribe the handsets.

Note: The CDCS command can also be used on MCMO type sets.

The CDCS conversion routine prints each TN as it is moved, in the following format:

500 TN l s c 00 = DCS TN L' S' C' Index#.
where:

L' S' C' = virtual TN
Index# = default of the unit number of the 500 type set.

LD 20 – Print actual DMC TN and virtual DMC TN list.

Prompt	Response	Description
REQ	PRT	Request.
TYPE	DCS	Digital Cordless Set.
TN	 l s c u c u l s c u c u	Terminal Number for actual DMC For Large Systems For Small Systems and Succession 1000 systems Virtual Terminal Number For Large Systems For Small Systems and Succession 1000 systems

The print routine outputs the following format:

INDX Index # VTN lll s cc uu
where:

Index # = Index number of virtual TN.
lll s cc uu = Virtual TN of unit.

LD 81 – Print DCS features.

Prompt	Response	Description
REQ	LST	Request.
FEAT	VSIT	Feature Request - DECT visitors.
HMDN	Xx / <cr>	HoMe Directory Number. Specify a single HMDN or print all HMDN on system.

The LD 81 output format as follows:

DCS Cust# Local DN TN lll s cc uu HMDN Home DN Last Activity Date.
where:

Cust# = Customer Number
Local DN = Local Directory Number of user
lll s cc uu = TN of unit
Home DN = Home directory number of user
Last Activity Date = Last date of service change activity for user

LD 83 – Prints DCS terminal numbers with a unit type of DCS instead of 500.

Feature operation

To activate the MSMN feature:

1 Turn the handset on within the coverage range of a visited Companion DECT system.

2 Enter the coverage range of a visited Companion DECT system from another Companion DECT system with the handset on.

To deactivate the MSMN feature:

1 Turn the handset off within coverage range of the visited Companion DECT system. (The handset must have the DECT Detach feature.)

2 Turn the handset on at the home Companion DECT system. (Any CFW related to the handset cancels.)

3 Enter the coverage range of the home Companion DECT system with the handset on. (Any CFW related to the handset cancels.)

Multi-Tenant Service

Contents

This section contains information on the following topics:

Feature description 1111
Operating parameters 1118
Feature interactions 1120
Feature packaging 1130
Feature implementation 1130
Feature operation 1135

Feature description

The Multi-Tenant Service feature enables customers to resell system features and services to other users. The stations belonging to the customer can be divided into customer sub-groups known as tenants. Tenants are separated by programming access restrictions on a tenant-by-tenant basis.

Administrators can configure access to other tenants, Attendant Consoles, and trunk routes so that tenants have private access to some services and shared access to others. Multi-Tenant Service can also be configured to deny access to certain services. Records of tenant activity are maintained through Call Detail Recording (CDR).

Telephones that are not assigned tenant status belong to one of the customers allowed. These customer resource telephones have access to all other telephones, Attendant Consoles, and outgoing trunk routes belonging to the same customer.

The number of tenants that can be configured on a per customer basis is dependant on the number of configured customers and the amount of available memory. The maximum number of tenants is 512 per customer.

Tenants receive all the features defined by the customer. Features that are handled at the tenant level include:

- Incoming Call Indicators
- Call Waiting Indicator
- Recorded Overflow Announcement
- Listed Directory Numbers
- Attendant Overflow Position
- Night Directory Number

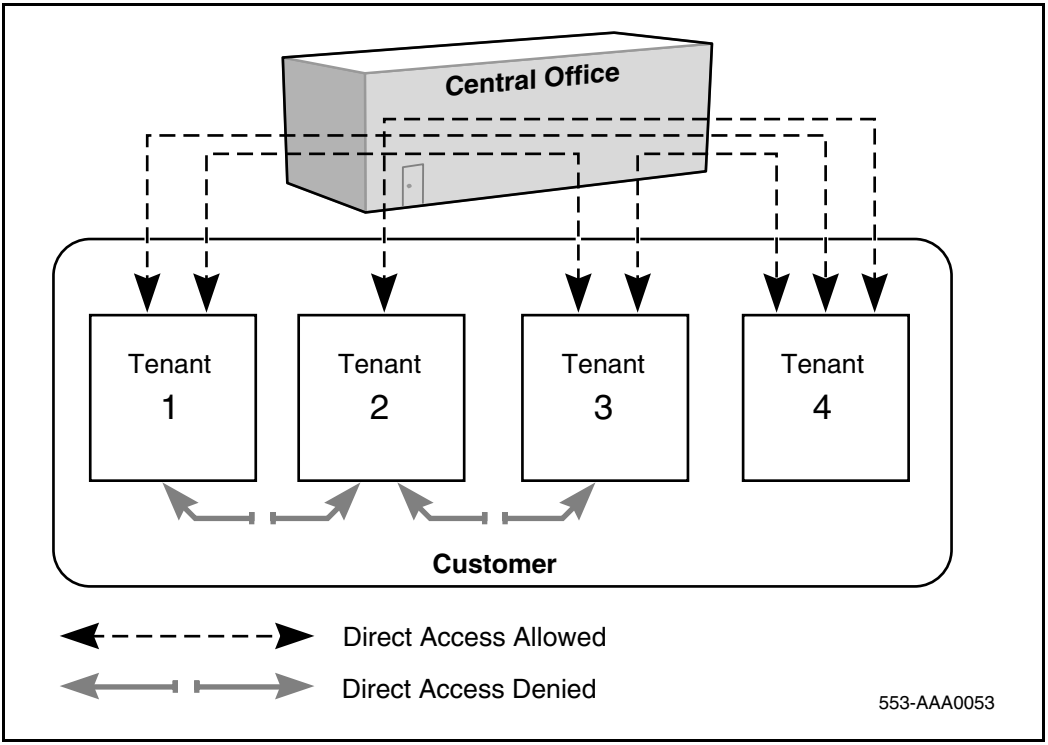
Tenants share the same numbering plan as their service provider. The following capabilities are defined on a tenant-by-tenant basis:

- Tenant-to-Tenant Access
- Tenant to Trunk Route Access
- Tenant to Attendant Console Grouping

Tenant-to-Tenant Access

Calls between tenant groups for the same customer are defined by Tenant-to-Tenant Access. As shown in Figure 72 on page 1113, a tenant is configured to allow or deny direct internal call access to some or all tenants of the same customer. To reach these tenants, the caller must dial the tenant's Listed Directory Number (LDN) through the Central Office. Access is always two-way. Therefore, if Tenant A has direct internal call access to Tenant B, Tenant B also has direct internal call access to Tenant A. Customer telephones not belonging to a tenant have two-way access to all tenant telephones in the customer group.

Figure 72
Tenant-to-Tenant access



As shown in Table 87, Tenant-to-Tenant Access allows or denies tenants of the customer:

Table 87
Tenant-to -Tenant Access allowed or denied

Tenant	Direct access allowed	Direct access denied
1	3 and 4	2
2	4	1 and 3
3	1 and 4	2
4	1, 2, and 3	

Outgoing Tenant-to-Trunk Route Access

Tenant-to-trunk route access applies only to outgoing calls. All tenants have access to incoming calls on any route. Customer telephones have access to all the customer's outgoing routes.

A tenant can have private outgoing trunk routes assigned. This is done by denying all other tenants access to the routes. Figure 73 on page 1115 shows a diagram of the following tables.

Table 88
Tenant Access to Private Routes

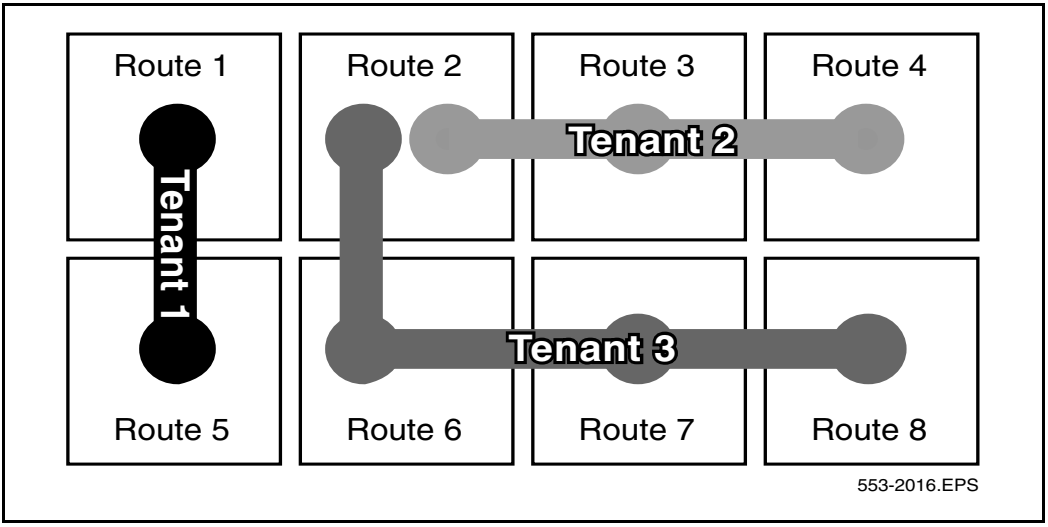
Tenant	Private Access Route
1	1 & 5
2	3 & 4
3	6, 7 & 8

A tenant can share outgoing trunk routes with other tenants of the same customer. As shown in Figure 73 on page 1115, Tenants 2 and 3 share access to route 2.

Table 89
Tenant Restrictions to Outgoing Routes

Tenant	Restricted Access to Trunk Routes
1	2, 3, 4, 6, 7 & 8
2	1, 5, 6, 7 & 8
3	1, 3, 4 & 5

Figure 73
Tenant-to-Trunk Route Access



Tenant to Attendant Console Grouping

With Multi-Tenant Service, all Attendant Consoles are placed into groups which are associated with specific tenants and specific incoming trunk routes. The Group Number range is 0 to 511. All Attendant Consoles configured for a customer are automatically members of group 0. The other groups are defined in the software to fit tenant requirements. For more information, refer to “Tenant to Attendant Console Grouping” on page 1115

Tenant-to-Attendant Access (Internal Calls)

Tenant-to-Attendant Access specifies which Attendant Console Group receives automatic presentation of a tenant's dial-zero calls.

Trunk Route-to-Attendant Access

Route-to-Attendant access specifies which Attendant Console Group receives automatic presentation of incoming calls from a particular route.

Console Presentation Groups

Console Presentation Groups (CPGs) are assigned to handle attendant calls from one tenant for a customer or for calls originating from certain trunks in a particular route.

Most Attendant Console features and parameters apply to CPGs. For more information, refer to “Attendant Console features” on page 1116.

Attendant Console features

Internal attendant-DN calls

When a tenant telephone dials the attendant DN, the call is presented to an idle Attendant Console. The call is routed to an Attendant Group associated with the tenant of the calling telephone, if Attendant Console Groups have been specified for the tenant. Otherwise, calls are presented to any idle Attendant Console belonging to the customer. For example, in Table 90 on page 1118, an attendant DN call from a Tenant 2 telephone is presented to an idle attendant in group 2 (console 1 or 2).

Incoming external calls

Incoming external calls are presented only to the Attendant Console Group specified to serve the trunk group. Also from Table 90, incoming calls on route 3 are presented to Attendant Consoles in group 6 (console 9 or 10).

Attendant Initiated Calls

All attendants have access to the customer’s numbering plan and can initiate a call to any customer tenant.

Attendant Overflow Position (AOP)

The Attendant Overflow Directory Number (AODN) should be accessible to all tenants. Attendant calls from tenants who do not have AODN access will not divert to AODN. They remain in the attendant queue.

Attendant Recall

When a tenant telephone recalls the attendant, the call is presented to an attendant in a group specified for the tenant of the calling telephone.

Attendant Extended Call

When an attendant extends a call from tenant A to tenant B, a 3-way conversation is set up only if tenant A and tenant B are allowed Tenant-to-Tenant Access.

Automatic Timed Recall (ATR)

When Automatic Timed Recall (ATR) alerts the attendant, the call is presented to an attendant within the Tenant group of the originally called number.

Console Presentation Group

A Console Presentation Group (CPG) is a subset of all consoles configured for a customer. A CPG is assigned to handle attendant calls from one tenant for a customer, or calls originated by trunks on a route.

CPG improves functions on the following CPG Level Services:

- **Attendant Overflow Positions** Each CPG can have its own AOP-DN and waiting time threshold.
- **Call Waiting Indication** The count thresholds, timers and buzz options for Call Waiting are defined for each CPG.
- **Incoming Call Identification** The ICI keys are defined for each CPG. Attendants see only those ICI key definitions for their own CPG.
- **Listed Directory Numbers** Each CPG allows four LDNs.
- **Night Service** Each CPG can go into Night Service mode regardless of the status of the other CPGs.

Access to incoming trunk routes

Any tenant can access an incoming call from any incoming trunk route. Attendant Console Groups can be specified to receive automatic presentation of incoming calls from specified routes. This includes calls that terminate at an Attendant Console and calls that intercept to an Attendant Console. For example, as seen in Table 90, incoming calls on route 2 are automatically presented to Attendant Console Group 5 (console 7 only).

Table 90
Typical attendant group arrangement

Attendant group number	Attendant consoles	Incoming Trunk routes	Tenant
0	1-10		
1	1		1
2	1, 2	1	2
3	1		3
4	3, 4	4	
5	7	2, 5	
6	9, 10	3	

Access to outgoing trunk routes

Tenants dial the appropriate trunk route Access Code to connect to a trunk route. Access Codes are assigned on a trunk route basis. Therefore, all tenants use the same Access Code to connect to a particular route. Customer telephones have access to all outgoing trunk routes belonging to their customer. Access to specific trunk routes is allowed or denied to individual tenants through service change. Tenants who attempt to access denied routes receive normal intercept treatment.

Operating parameters

Multi-Tenant Service is not supported by Meridian Mail applications.

Traffic data is collected on a per customer basis only.

Tenants can receive private or shared access to the Modem Trunk routes configured for their customer.

All tenants have access to their customer's Music trunks.

Tenants can receive private or shared access to the Paging routes configured for their customer.

All tenants can access their customer's recorded Announcement (RAN) trunks.

Individual tenants can be allowed or denied trunk access (private or shared) for the following trunk types:

- Add-on Data Module
- Centralized Automatic Message Accounting
- Common Controlled Switching Arrangement
- Central Office
- Direct Inward Dialing
- Dictation trunk
- Direct Outward Dialing
- Foreign Exchange
- Modem
- Paging trunk
- TIE
- Wide Area Telephone Service.

There are no restrictions on calls routed to the following trunk types:

- Automatic Identification of Outward Dialing
- Music trunk
- Recorded Announcement
- Release Link
- Main
- Release Link
- Remote Emergency Recorder

Feature interactions

Access Restrictions

Multi-Tenant Access Restrictions affect the way that tenants interact with other tenants, trunk routes, and Attendant Consoles.

In general, Multi-Tenant Access Restrictions take precedence over the system features with which they interact.

For example, when a direct Tenant-to-Tenant call has been made, the called party cannot transfer the call to a different tenant if the first and third tenants are denied access to each other.

In addition to Class of Service and Trunk Group Access Restrictions (TGAR) and Trunk Access Restriction Group (TARG) restrictions, Multi-Tenant Service can impose the following access restrictions:

- Tenant-to-Tenant
- Tenant-to-Trunk Group
- Tenant-to-Attendant Group
- Trunk Group-to-Attendant Group

Attendant Administration

An Attendant can dial the Access Code and activate the Administration Mode for that CPG group. In this mode, attendants can modify the configuration of any set for this customer.

Automatic Timed Recall

When Automatic Timed Recall (ATR) alerts the attendant and Multi-Tenant Services are in effect, the call is presented to an attendant in the same tenant group as the originally dialed DN.

Basic Authorization Codes

All tenants share their customer's Authorization Code tables. However, Tenant-to-Tenant and Tenant-to-Trunk Route specifications override Basic Authorization Codes (BAUT).

Call Detail Recording

With Multi-Tenant Service, all tenants are included in CDR records. The tenant numbers of the originating and terminating parties are added to the CDR records as shown in Table 91.

Table 91
CDR record types and descriptions

CDR record type	Description
A	Authorization Code
C	Charge Account
E	End
L	Internal Record
M	Charge Conference
N	Normal
P	Calling Party Number
Q	Connect Record
S	Start

Tenant and customer numbers are included by the system in the CDR output to provide the customer with data for call billing and chargeback activities.

Call Forward All Calls

Originating Party COS

If the calling party (CFO) option is defined in the Customer Data Block (LD 15), inter-tenant Call Forward is allowed if the calling party's tenant has access to the Call Forward DN tenant and the dialed DN tenant. If the Call Forward DN is in a tenant group that the caller cannot access, the DN is treated as invalid, and the caller receives an overflow tone. The software performs an access check.

Forwarding Party COS

If the forwarding party (CFF) option is defined in the Customer Data Block (LD 15), inter-tenant Call Forward is allowed if the Call Forwarding party's tenant has access to the tenant of the Call Forward DN. The local Telephone Company decides whether the option is available.

Call Forward Busy

DID calls to a busy telephone are forwarded to an idle Attendant Console specified for the tenant of the dialed telephone.

Hunting and Call Waiting take precedence over Call Forward Busy.

Call Forward No Answer

Attendant option

After a customer-defined number of rings, an unanswered call forwards to an idle Attendant Console specified for the tenant of the dialed telephone.

Any DN option

If the tenant of the calling party has access to the tenant of the Call Forward DN, the unanswered call forwards to the Call Forward DN. If Tenant-to-Tenant Access is denied, the call is processed as if no CFNA-DN exists.

Secretarial Filtering

Calls receive Secretarial Filtering only if the tenant of the Call Forward DN is accessible by the tenant of the caller.

Call Forward No Answer, Second Level

All of the same operations apply to the forwarded DN when Second Level CFNA is allowed.

Call Forward by Call Type

The originally dialed DN must have access to the tenant of the forwarding DN. This allows external calls to easily forward to the programmed DN.

To forward an internal call by CFCT, the originator must have access to the tenant of the programmed forwarding DN.

Call Park

Parked calls recall back to the Attendant who parked them. If that attendant goes into Position Busy mode, then the Parked call recalls to an attendant in the same CPG as the original. Recalls to Attendants going into Night Service mode return to the attendant queue until the caller abandons the call.

Tenant access checking between set (A) who picks up a parked call, and party (B), who parked the call, is enforced as follows:

- If B is a set, tenant-to-tenant access must be allowed between A and B.
- If B is an attendant, A and B must belong to the same CPG for tenant-to-tenant access.
- If access is denied, set A (who intends to pick up the access-denied parked call) receives a blocking tone.

Call Transfer

A telephone user can transfer its original party to a third party only if the transferred parties can access each other. Software prevents joining tenants who are denied access to each other.

Calls Waiting Indication

The Calls Waiting Indication displays the calls waiting count for the customer. It is not tenant related, but because routes and tenants specify the consoles to which calls are automatically presented, a non-zero call waiting count can be displayed. This occurs even though no calls are presented to the console.

Centralized Attendant Service

Specific Attendant Consoles can be assigned to receive automatic presentation of incoming calls from Release Link-Main (RLM) trunks.

All tenants have access to Release Link-Remote (RLR) trunks.

Code restriction

The code restriction data configured for a customer, applies to all tenants belonging to that customer.

Conference

All members of a conference must have access to each other. Large System software runs an access check which prevents the addition of access denied tenants.

Controlled Class of Service

The tenant of the Controlled Class of Service Controlling Station must have access to the tenant of the controlled telephone to activate CCOS.

Departmental Listed DN

The Departmental Listed Directory Number (DLDN) takes precedence over Multi-Tenant Service. For either Dial-Zero or Recall, initiated from a tenant telephone, two events may occur. First, the call is presented to the DLDN attendant when the telephone has specified DLDN. Second, the call is presented to the console specified by the telephone's tenant when the telephone does not have DLDN specified.

Dial Intercom Group

The tenant of the dialing telephone must have access to the tenant of each telephone reached by Dial Intercom Group (DIG) dialing.

Electronic Switched Network

All tenants have access to the Electronic Switched Network (ESN) features specified at the customer level. Except for Tenant-to-Route access, all ESN features are identical for each tenant belonging to the same customer.

Coordinated Dialing Plan

All tenants can access the complete Coordinated Dialing Plan (CDP) if they are configured for access to TIE trunk routes that are a part of the CDP.

Flexible Call Back Queuing

The originating tenant must have access to an eligible route in the Call Back Queue (CBQ) route list.

Free Calling Area Screening

Free Calling Area Screening checks occur normally if the originating tenant has access to the selected route.

Basic Alternate Route Selection
Network Alternate Route Selection

All tenants have access to the BARS/NARS Access Codes of their customer. Tenants that do not share access to the selected route are denied access to that route.

Network Authorization Code

Network Authorization Code (NAUT) does not override Tenant-to-Route Access restrictions within the call originator's Large System.

Network Speed Call (NSC)

All tenants have access to their customer's NSC lists. Any route selected by NSC must have Tenant-to-Route Access allowed.

Off-Hook Queuing (OHQ)

OHQ is allowed if the tenant has access to a route in the initial route list of their customer that is eligible for OHQ.

Flexible Hot Line

Flexible Hot Line allows designated telephones to place calls to a predetermined destination by going off-hook. If the Hot Line telephone's tenant does not have access to the tenant of the Hot Line DN, standard intercept treatment is provided.

Group Call

Group Call allows a Meridian 1 proprietary set user to place a call to a maximum of 10 (maximum of 6 for Small Systems) predefined DNs simultaneously by pressing a Group Call key. The tenant of the telephone initiating the Group Call must have access to the tenant of each member in the group. Restricted members are excluded from the group. The system undertakes access checks comparing the originator against each group member.

Hunting

Circular, Linear, Secretarial or Short Hunting routes call from a busy DN to the next idle DN in a prearranged group. If the hunted DN being hunted is not accessible to the dialing telephone, it is handled as an invalid member in the hunting chain. Short Hunting requires that all DNs configured on a QSU telephone belong to the same tenant.

Hunting Route

One step Route Hunting takes place between routes of the same trunk type. Tenants share their customer's route hunting specification and can use the stepped to route if they have Tenant-to-Route Access allowed for the route.

Integrated Messaging System (IMS)

Tenants can share or be denied access to their customer's IMS.

Integrated Voice Messaging System (IVMS)

Tenants can share or be denied access to their customer's Integrated Voice Messaging Service (IVMS). Tenants who do not have direct access to each other can use the IVMS Broadcast capability to leave messages for each other.

Intercept Treatment

All tenants share the customer's intercept specification.

When Tenant-to-Route Access restricts a Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) call, intercept treatment is the same as any invalid BARS/NARS call.

When an internal call intercepts to an attendant because of defined restrictions or dialing irregularities, it automatically presents the call to one of the Attendant Consoles specified for the calling tenant.

When intercept treatment includes a Recorded Announcement (RN) and Tenant-to-Tenant Access restricts a call, an Access Denied RAN plays.

Field Lamp Array

The Lamp Field Array, located on either an Attendant Console or a QSU telephone, indicates the busy/idle status of 150 consecutive DNs. These DNs display regardless of Tenant-to-Tenant Access specifications of the array equipped tenant telephone. For this reason, the DNs assigned in the array should be accessible by the tenant of the array associated telephone.

Maintenance telephone

QSU telephones with maintenance allowed COS must receive access to all tenants, all trunk routes, and all Attendant Consoles.

Manual service

When a manual telephone goes off-hook, the call is presented to an idle Attendant Console belonging to a group specified for its tenant.

Manual Trunk service

When an incoming trunk terminates on a DN, there is no access check. Incoming trunks terminate on an Attendant Console only if the console is specified for that manual trunk route.

Tenant-to-Route access checking is completed for outgoing manual trunk calls.

Multiple Appearance DNs

All appearances of a DN should reside on telephones belonging to the same tenant. When a multiple appearance DN is called, the last non-fully restricted Terminal Number (TN) in its TN list determines the terminating tenant number for Tenant-to-Tenant Access checking.

Multiple Listed Directory Numbers (MLDN)

Route-to-Attendant Console Access determines which Attendant Console Group receives automatic presentation of calls from a specific Direct Inward Dialing (DID) trunk route. Each of the four DID LDNs are configured to have its calls presented at the loop key of specific Attendant Consoles by using DLDN.

Night Service

Automatic Call Distribution (ACD) allows special functionality for the system under certain conditions, such as Night Service.

The Night DN should be assigned as a customer resource so all tenants have access to the Night DN for internal calls when Night Service is in effect. Otherwise the call is treated as if no Night DN exists.

Position Busy

When all Attendant Consoles designated to receive incoming trunk calls from a particular trunk route are in Position Busy, incoming trunk calls from those routes are directed to the Trunk Night Service DN.

Office Data Administration System

Office Data Administration System (ODAS) does not contain tenant information.

Ring Again

Ring Again is permitted when the originating tenant has access to the destination tenant.

Ringing Number Pickup

Ringing Number Pickup (RNPU) enables a telephone to answer calls to other telephones in the same RNPU Group. All tenants have access to their customer's RNPU Access Code. Members of an RNPU group can only answer calls for other members if their tenant has access to the tenant of the calling party. For this reason, members of an RNPU group are selected from telephones belonging to the same tenant. The calling party's access is checked against the called party by the system.

Route Selection-Automatic Number Identification

All tenants can dial the Route Selection - Automatic Number Identification (RS-ANI) DN. The ANI route selected from the RS-ANI list is used only if the tenant of the originating telephone has access to the route.

Secrecy

The Secrecy option, specified for a customer, applies to all CPG attendants for that customer.

Speed Call

Speed Call allows a telephone user to place calls to specified DNs by dialing a two-digit code. A user of a Speed Call List receives normal intercept treatment if the tenant does not have access to the listed destination tenant.

Supervisory consoles

Supervisory consoles specified for a customer belong to one Console Presentation Group (CPG). In the Supervisory mode, ICI lamps show only the information for ICIs in that CPG. The thresholds specified in the Customer Data Block apply only to the CPG where that console resides, and they do not affect any other CPG.

System Speed Call

All tenants share their customer's System Speed Call (SSC) lists. When a System Speed Call DN is used Tenant-to-Trunk Route access restrictions apply.

Trunk Group Access Restrictions

All tenants share their customer's Trunk Group Access Restrictions (TGAR), but Tenant Service Access restrictions take precedence, even though the telephone COS and TGAR do not restrict access to a route. Normal intercept treatment is provided when Tenant Service Access is denied.

Trunk routes
Voice Call

Tenant-to-Tenant Access must be allowed between the Voice Call originating telephone and terminating telephone.

Feature packaging

The following packages are required for Multi-Tenant Service:

- Multi-Tenant Service (TENS) is package 86, which requires:
  - — Console Presentation Groups (CPGS) package 17.

Other features expected in a Console Presentation Group environment must be packaged for complete functionality. They are as follows:

- Centralized Attendant Service-Remote (CASR) package 26
- Centralized Attendant Service-Main (CASM) package 27
- Recorded Overflow Announcement (ROA) package 36
- Attendant Overflow Position (AOP) package 56

The maximum number of route list entries for BARS/NARS is always 64.

CPG services are mutually exclusive with Departmentally Listed DNs (DLDN).

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 93 – Enable, disable, or print Multi-Tenant Service for a specified customer.

2 LD 93 – Allow, deny, or print Tenant-to-Tenant Access for a specified tenant.

3 LD 93 – Allow, deny, or print Tenant-to-Route Access for a specified trunk route.

4 LD 93 – Add Console Presentation Group.

5 LD 93 – Assign Tenant-to-Attendant Console access.

6 LD 93 – Assign Attendant Console group number.

7 LD 10 – Add Multi-Tenant Service assignments on analog (500/2500) telephones.

8 LD 11 – Add Multi-Tenant Service assignments on proprietary telephones.

LD 93 – Enable, disable, or print Multi-Tenant Service for a specified customer.

Prompt	Response	Description
REQ	NEW OUT PRT	Add, remove, or print.
TYPE	TENS	Tenant service data block.
CUST	xx	Customer number, as defined in LD 15
TEN	1-511	Tenant Number.

LD 93 – Allow, deny, or print Tenant-to-Tenant Access for a specified tenant.

Prompt	Response	Description
REQ	CHG PRT	Change or print.
TYPE	TACC	Tenant-to-Tenant Access Data Block.
CUST	xx	Customer number, as defined in LD 15
TEN	1-511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service.
ACC	DENY ALOW	Tenants denied access are to be entered. Tenants allowed access are to be entered.
DENY	1-511 1-511 ALL	Tenant numbers denied access to and from this tenant (prompted if ACC = DENY). All tenant numbers denied access to and from this tenant (tenant can only access itself).
ALOW	1-511 1-511 ALL	Tenant numbers allowed access to and from this tenant (prompted if ACC = ALOW). All tenant numbers allowed access to and from this tenant.

LD 93 – Allow, deny, or print Tenant-to-Route Access for a specified trunk route.

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	RACC	Tenant-to-Route Access Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
ACC	DENY ALOW	Tenants denied access to the route are to be entered. Tenants allowed access to the route are to be entered.
DENY	1-511 1-511 ALL	Tenant numbers denied access to this route (prompted if ACC = DENY). All tenant numbers denied access to this route.
ALOW	1-511 1-511 ALL	Tenant numbers allowed access to this route (prompted if ACC = ALOW). All tenant numbers allowed access to this route

LD 93 – Add Console Presentation Group.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	CPG	Console Presentation Group data block.
CUST	xx	Customer number, as defined in LD 15
AGNO	1-63	Attendant Console group number. Attendant Console group 0 (AGNO 0) always exists and contains all Attendant Consoles configured for the customer.
ANUM	1-63 1-63	Add attendant console numbers.

LD 93 – Assign Tenant-to-Attendant Console access.

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	TCPG	Tenant -to-Attendant Console Group data block.
CUST	xx	Customer number, as defined in LD 15
TEN	1-511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service.
AGNO	0-63	Attendant Console group number.

LD 93 – Assign Attendant Console group number.

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.
TYPE	RCPG	Route-to-Attendant Presentation Group data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
AGNO	0-63	Attendant Console group number.

LD 10 – Add Multi-Tenant Service assignments on analog (500/2500) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(TEND) TENA	Tenant service (denied) (station shares customer resources and is a non-tenant). Tenant service allowed.
TEN	1-511	Tenant number (prompted if CLS = TENA). Tenant 0 is reserved for telephones with a TEND Class of Service.

LD 11 – Add Multi-Tenant Service assignments on proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(TEND) TENA	Tenant service (denied) (station shares customer resources and is a non-tenant). Tenant service allowed.
TEN	1–511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service. Prompted if CLS = TENA.

Feature operation

No specific operating procedures are required to use this feature.

Music

Contents

This section contains information on the following topics:

Feature description 1137
Operating parameters 1138
Feature interactions 1139
Feature packaging 1140
Feature implementation 1141
Feature operation 1143

Feature description

The Music package supports Music on Hold and Automatic Call Distribution (ACD) Music on Delay. One or more music sources can be connected to one or more music trunks on peripheral equipment. Each music trunk is assigned to a music route and to a conference loop. Incoming callers are bridged into a listen-only conference and provided with music when on hold or when waiting for an ACD call to be answered.

Music on Delay

Music on Delay presents a listen-only path to a music source for calls waiting in ACD queues. Music on Delay sources are identified separately for each Automatic Call Distribution Directory Number (ACD DN). Complete details are described in *Automatic Call Distribution: Description* (553-3001-351).

Music on Hold

This feature allows incoming calls over a CO, FX, WATS, DID, or TIE trunk to receive music if placed on hold. Music is provided only if the trunk route is defined to receive music. The trunks selected to receive music are provided with a listen-only path to a music conference connection.

Music is provided by a dedicated music trunk by means of the conference circuit. To minimize blocking of the music conference, at least two conference loops must be assigned in each network group requiring music. The loop with the higher number should not be assigned to music trunks.

To activate the Music on Hold feature on virtual trunks, the following are required:

- Install package 328 (MUSBRD).
- Set the BDCT prompt in LD 16 to YES.

Operating parameters

Music is provided by a Recorded Announcement (RAN) or universal trunk circuit card.

Only trunks assigned to a route specified by service change receive Music on Hold.

When a call is held, the system looks for a network path to provide the music. If a path is not found, no music is heard.

When a Universal trunk card is used, Music and RAN trunks can be assigned to the same card.

Connections blocked once are not automatically attempted again.

Simple source-only connections on the Attendant Console receive music; all others do not.

Main Release Link Trunks do not receive music.

Calls to special trunks (such as Paging or Dictation) do not receive music if placed on hold.

The music trunk Terminal Number (TN) must be within the same network group as the conference circuit to which it is assigned.

One music trunk per customer must be located in each network group requiring music.

Music is not supplied across groups. For example, if group 4 does not have a music trunk and groups 0-3 have music trunks, an incoming call to group 4 placed on hold will not receive music.

A single conference loop with one music trunk assigned can support up to 29 simultaneous listeners.

If more than one music trunk is assigned to one conference loop, they must use different routes. The total number of possible listeners is 30 minus the number of assigned trunks. Additional music trunks and conference loops can be configured if required.

The music source must be compatible with the music trunk circuit pack.

Feature interactions

AC15 Recall: Transfer from Norstar

A party put on hold by an AC15 trunk will hear music if it is configured.

Attendant Trunk Group Busy Indication

A music route that appears on a Trunk Group Busy key on the Attendant Console cannot be controlled by activation of the Trunk Group Busy key. In addition, the associated lamp will not reflect the status of the music trunks.

Break In with Secrecy

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Call Park

When a call is parked, music is not heard. When a trunk is parked, music plays if music is enabled for the route.

Conference

With basic Music on Hold, when a call is placed on consultation hold while a Conference is being established, music does not play. Enhanced Music (EMUS) package 119 is required for music on consultation hold (see "Music, Enhanced" on page 1145).

Group Hunting Queuing Limitation

No music is provided for Group Hunting Queuing Limitation.

On Hold on Loudspeaker

Music on Hold is not be heard by either party during a loudspeaker call.

Recovery on Misoperation of Attendant Console

Music on Hold is applied to calls put on hold due to AUTOHOLD.

Source Included when Attendant Dials

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Trunk Traffic Reporting Enhancement

The Trunk Seizure Option is not supported on a music trunk.

Feature packaging

The Music feature requires the following packages:

- Music (MUS) package 44
- Recorded Announcement (RAN) package 7

To configure Music on Delay for an ACD environment, the following packages are also required:

- Base Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution, Package A (ACDA) package 45
- Automatic Call Distribution, Package B (ACDB) package 41

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Enable conference loops for Music on Hold.

2 LD 16 – Enable a music route.

3 LD 14 – Enable a music trunk.

4 LD 16 – Enable Music on Hold for trunk routes.

5 LD 23 – Enable Music for an Automatic Call Distribution Directory Number.

LD 17 – Enable conference loops for Music on Hold.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN CEQU	Configuration Record. Gate opener.
CEQU	(NO) YES	Change to CE parameters.
- XCT	0-158	Loop number for NT8D17 Conference/TDS/MFS card. Enter an even network loop number for TDS/MFS functions. The conference function is automatically assigned the next higher (odd) loop number.
- CONF	0-158	Loop number for conference card.

LD 16 – Enable a music route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
TKTP	MUS	Music route.
ICOG	OGT	Outgoing route only.
ACOD	xxxx	Trunk route access code.
Note: All other prompts can be set to default values.		

LD 14 – Enable a music trunk.

Prompt	Response	Description
REQ	NEW	New.
TYPE	MUS	Music trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
CFLP	0-158	Conference loop assigned to music in LD 17.

LD 16 – Enable Music on Hold for trunk routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	COT DID FEX TIE WAT	Route type.
MUS	(NO) YES	Music on Hold (is not) or is to be provided for this trunk route.
MRT	xxx	Music route number.

LD 23 – Enable Music for an Automatic Call Distribution Directory Number.

Prompt	Response	Description
REQ	NEW	Add.
TYPE	ACD	Update the ACD data block.
CUST	xx	Customer number, as defined in LD 15
ACDN	xxx...x	ACD DN.
MURT	X 0-511	Music route number. X = remove route.

Feature operation

No specific operating procedures are required to use this feature.

Music, Enhanced

Contents

This section contains information on the following topics:

Feature description 1145
Operating parameters 1146
Feature interactions 1146
Feature packaging 1147
Feature implementation 1148
Feature operation 1150

Feature description

Enhanced Music (EMUS) provides music for internal and external calls. Music is provided when telephones are placed on Hold, Consultation Hold, and Camp-On and when calls at the Attendant Console are split using the “Exclude Source/Destination” keys.

Enhanced Music (EMUS) provides music in situations described in Table 92.

Table 92
Features vs. no Music, Music, and Enhanced Music

	Without Music		Music		Enhanced Music	
	Sets	**Trunks**	**Sets**	**Trunks**	**Sets**	**Trunks**
ROA Waiting	No	No	Yes	Yes	Yes	Yes
Call Park	No	No	Yes	Yes	Yes	Yes
ACD Music	No	No	Yes	Yes	Yes	Yes
Hold Key	No	No	No	Yes	Yes	Yes
Permanent Hold	No	No	No	Yes	Yes	Yes
Consultation Hold	No	No	No	No	Yes	Yes
Splitting	No	No	No	Yes	Yes	Yes
Camp-On	No	No	N/A	Yes	N/A	Yes

Operating parameters

The requirements for Enhanced Music on Hold are the same as for Music on Hold. See “Music” on page 1137.

Trunks receive Music on a route basis. Telephones receive Music on a customer basis.

Feature interactions

Enhanced Music on Hold has the same feature interactions as Music on Hold. In addition, it has interactions with the following features:

Attendant Busy Verify

When the attendant attempts to Busy Verify a telephone receiving Music, the Music is removed. When the attendant releases, Music is returned.

Call Hold, Deluxe

A caller placed on Hold by a member of a multiple appearance group receives Music regardless of whether the call is on Hold or Exclusive Hold.

Call Transfer

The held party receives Music when the other party presses the Call Transfer key. The Music connection remains until the Call Transfer key or the DN key is pressed, ending the Consultation Hold state.

Charge Account and Calling Party Number

The Charge Account (CHG) and Calling Party Number (CPN) keys place the far end party on Hold while a charge number is entered. The held party receives Music during this period.

Conference

The held party receives Music when the Conference key is pressed, while the conference is being established, and whenever the conference is reduced to two parties with one party on Hold. Once the conference is established, Music is no longer provided.

A Six-party Conference operates the same as a Three-party Conference.

Privacy Release

When using Privacy Release to add one or more members to a call already receiving Music, the Music is removed.

Telephones - M3000

The Switch Parties key allows Music to the party on Hold and ends Music to the other party each time it is pressed.

Feature packaging

Enhanced Music (EMUS) package 119 requires:

- Music (MUS) package 44, and
- Recorded Announcement (RAN) package 7.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Add or change Conference loops for Music on Hold.

2 LD 15 – Enable Music Customer Data Block.

3 LD 16 – Enable a Music route.

4 LD 14 – Enable a Music trunk. At least one Music trunk per network group is required for each customer requiring Music.

5 LD 16 – Enable Music on Hold for trunk routes.

LD 17 – Add or change Conference loops for Music on Hold.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN CEQU	Configuration Record.
CEQU	(NO) YES	Change to CE parameters.
- XCT	0-158	Loop number for NT8D17 Conference/TDS/MFS card. Enter an even network loop number for TDS/MFS functions. The conference function is automatically assigned the next higher (odd) loop number.
- CONF	0-158	Loop number for conference card (must be an even numbered loop).

LD 15 – Enable Music Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CDB FTR	Customer Data Block Features and options

CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- MUS	(NO) YES	Enhanced music for telephones.
- MUSR	0-511	Music route for telephones.

LD 16 – Enable a Music route.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
TKTP	MUS	Music route.
ICOG	OGT	Outgoing route only.
ACOD	xxxx	Trunk route access code.
Note: All other prompts can be set to default values.		

LD 14 – Enable a Music trunk. At least one Music trunk per network group is required for each customer requiring Music.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	MUS	Music trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
CFLP	0-158	Conference loop assigned to music in LD 17.

LD 16 – Enable Music on Hold for trunk routes.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
TKTP	COT DID FEX TIE WAT	Trunk type.
MUS	(NO) YES	Music on Hold (is not) is to be provided for this trunk route.
MRT	0-511	Music route number.

Feature operation

No specific operating procedures are required to use this feature.

Music Broadcast

Contents

This section contains information on the following topics:

Feature description 1151
Operating parameters 1155
Feature interactions 1157
Feature packaging 1158
Feature implementation 1158
Feature operation 1160

Feature description

The Music Broadcast feature expands existing Music functionality. This feature provides the following enhancements:

- Broadcast Capabilities
- Incremental Software Management limit
- Traffic Study Option

Broadcast Capabilities

Existing Conference-based Music features require that each Music trunk be assigned to a Music route and to a Conference loop. Incoming callers are bridged into a listen-only conference and provided with music while on call hold or call waiting in an Automatic Call Distribution (ACD) environment. The existing Conference-based Music features support intra-group music only. Therefore, each network group must be provided with its own Music trunk.

The Music Broadcast feature allows the system to broadcast music to several parties at one time using a single Music Broadcast trunk port. This feature supports Music on Hold (MOH). Music is now delivered using system software; hence, Conference hardware is not required. It is no longer necessary to share Conference resources with Conference features, such as Conference and Group Call. Music Broadcast supports both intra-group and inter-group music. Therefore, a Music trunk in each network group is not required.

A Music Broadcast call consists of several one-way connections from the Music trunk to each caller. The Music Broadcast feature reduces the number of timeslots required for callers to listen to music while on call hold or call waiting in an Automatic Call Distribution (ACD) environment. One timeslot is required to enable Music trunk broadcasts. In addition, each party listening to music through the broadcasting music trunk requires one broadcast connection. The extra speech path resources that are needed for the existing Conference-based Music are unnecessary for Music Broadcast.

Incremental Software Management

An Incremental Software Management (ISM) limit is introduced for the Music Broadcast feature. This limits the total number of Music Broadcast connections allowed on a system. The ISM limit can be allocated over different Music routes and trunks. ISM allows a total of 64 Music Broadcast connections on one trunk at one time. If one trunk is configured with 64 connections, when the limit is reached, the 65th caller hears silence. The Music trunk is no longer available until a call disconnects. When a call disconnects, the next caller receives one of the Music Broadcast connections and receives music. However, the 65th caller still hears silence, even though a connection has become available.

If a customer has 64 connections configured on one trunk but requires more connections, additional trunks can be added to their system and additional connections can be purchased incrementally to a maximum of 9,999 connections. For example, should this customer require a total of 124 connections, an additional trunk and an additional 60 connections can be added to their original configuration. This provides the customer with a total of 124 connections (64 connections + 60 connections). LD 22 is modified to print the new ISM information for Music Broadcast connections. The existing SLT command prints the ISM information for the system.

The existing ISM header in LD 14 is modified to indicate the number of Music Broadcast connections allowed for the system. AVAIL shows the system's ISM limit for Music Broadcast connections. USED shows the number of configured Music Broadcast connections (the total number of Music Broadcast trunks for the system multiplied by the maximum number of connections per trunk). TOT shows the maximum number of Music Broadcast connections that can be supported on one system (AVAIL + USED).

The existing TN information shown in the ISM header in LD 14 is not modified by the Music Broadcast feature, as the amount of Music Broadcast trunk TNs is not checked against the ISM limit at SYSLOAD. The Music Broadcast ISM limit pertains to Music Broadcast connections only and not to TNs. Figure 74 is an example of the updated header:

Figure 74
ISM header in LD 14

TNS	AVAIL: xxxxx	USED: xxxxx	TOT: xxxxx
MUS CON	**AVAIL: xxxxx**	**USED: xxxxx**	**TOT: xxxxx**

Customers can modify ISM parameters via keycode. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects ISM parameters.

To expand ISM limits, customers must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in LD 143. For further information on keycode installation, please refer to *Large System: Upgrade Procedures* (553-3021-258).

For further information on ISM, refer to the "Incremental Software Management" on page 573 in this book.

Traffic Study Option

The Traffic Period Option (TPO) allows a customer to enhance their TFC002 reports to accumulate trunk usage data after every traffic period instead of accumulating usage only after a call disconnects. With this option enabled in LD 17, the Common Channel Signaling (CCS) associated with lengthy calls is reported in each traffic report interval throughout the duration of the call.

Previously, this feature did not apply to RAN and Music trunks. With the introduction of Music Broadcast, however, a Music call may last for an extended period of time. Therefore, changes are made to the Trunk Traffic Reporting Enhancement with the introduction of the TFC111 traffic report.

The TFC111 report provides information on the usage of broadcasting routes. For the TFC111 report to be output, customer report number 11 must be selected using the SOPC command in LD 2. For example, for Customer 0, SOPC 0 11 is entered. To print the TFC111 report, the TOPC command in LD 2 is used. For example, for Customer 0, TOPC 0 11 is entered. The TFC 111 report is also printed when automatic traffic reports are scheduled in LD 2.

The System Traffic message, TFS 0503, is output each time a music request cannot be completed because the total number of active Music Broadcast connections is equal to the system's ISM limit. Figure 75 is an example of the customer report, TFC111, for Music Broadcast routes.

Figure 75
TFC111 Report for a broadcasting music route

0200 (System ID)	TFC111
000 (Customer number)	
030 (Route number)	MUS (Trunk type)

Figure 75
TFC111 Report for a broadcasting music route

001132 (Successful broadcast connections peg count)	00016 (Average call duration)	00000 (Unsuccessful broadcast connections peg count)
00000 (Broadcast connections peg count for lowest usage trunk)	00000 (Broadcast connections peg count for second lowest usage trunk)	00002 (Broadcast connections peg count for third lowest usage trunk)

Operating parameters

Music Broadcast requires any Music trunk and an external music source **or** a Meridian Integrated RAN (MIRAN) card (NTAG36). MIRAN has the capability to provide audio input for external music.

A Conference loop is not required for Music Broadcast.

With the Music Broadcast package configured, both existing Conference-based Music and Music Broadcast can co-exist on the same system. The type of Music is dependant upon the BDCT prompt in the Route Data Block.

The Music Broadcast feature is applicable to Music routes only.

To upgrade an existing non-broadcasting Music route to a broadcasting Music route, the REQ prompt must be set to CHG and the BDCT prompt must be set to YES in LD 16.

A broadcasting Music route may only be changed to a non-broadcasting Music route if it is first removed in LD 16 and then added back into the system as a non-broadcasting Music route by setting the BDCT prompt to NO. In this case, the CFLP prompt in LD 14 must be defined, and the Conference loop number for the non-broadcasting Music route must match the loop number that was set previously in LD 17.

If more than 64 Music Broadcast connections are required due to high traffic, additional trunks, each with up to 64 Music Broadcast connections, can be added. This same Music source can be cross-connected to all Music trunk TNs within a particular Music Route.

When more than one Music trunk is attached to a broadcasting Music route, a trunk is first sought within the caller's own group. An already active trunk is chosen initially in order to give music to the requesting party. If there is not a Music trunk that is already active or if all active Music trunks already have the maximum number of callers connected, an idle trunk is sought. If an idle trunk is found, the call is connected. If there are no trunks available within the caller's group, trunks in other groups are sought.

Although Music Broadcast supports inter-group music, it is advisable that for multi-group systems with high inter-group traffic, a Music trunk be provisioned in each network group to reduce junctor traffic.

Several routes can be supported using Music Broadcast; hence, different types of Music are also supported. On multi-group systems, however, network group junctor traffic limitations may cause difficulty in supporting several types of music on one system. In this case, additional trunks and additional connections can be added to the system.

If blocking occurs, silence or ringback tone is given by the features requesting music.

When the actual number of Music Broadcast connections in use is equal to the ISM limit, another connection is not allowed. In this case, the Blocking operation is retained. Therefore, silence or ringback tone is given by the features requesting music. This information is output in the Traffic report (TFS 0503).

The total number of Music Broadcast trunks multiplied by the maximum number of Music Broadcast connections per trunk may be greater than the ISM limit. The ISM limit of Music Broadcast connections is shared between different types of Music routes.

When a Music Broadcast trunk port is forced to disconnect through maintenance, all connected callers hear silence but remain on hold.

Only those calls receiving music in an ACD queue are restored by the INIT ACD Queue Call Restore feature following a system initialization. Any other calls receiving music are dropped, and the callers hear silence.

Feature interactions

Call Detail Recording

Due to the number of callers that can be connected to a broadcasting Music trunk at one time, Call Detail Recording (CDR) is not supported on broadcasting Music routes.

> ***Note:*** CDR is prompted for Music routes in LD 16. However, even if CDR is set to YES, a CDR record will not be output for Broadcasting Music routes.

Integrated Call Center Management

The Integrated Call Center Management (ICCM) broadcast capability on a system is independent of the Music Broadcast capability which is applicable only to Music routes. This ICCM broadcast capability applies only to Interactive Voice Response (IVR) voice ports.

The script command GIVE MUSIC <music route number> connects a call to the specified Music route. The Music Broadcast feature is applied if appropriate.

The script command GIVE BROADCAST ANNOUNCEMENT {NOT INTERRUPTIBLE} <acd_dn> {WITH TREATMENT <treatment>} applies to IVR ports only, and the ICCM broadcast capability is applied in this case.

Meridian Interactive Voice Response

Interactive Voice Response (IVR) interacts with the Music Broadcast feature, using the existing functionality of a non-broadcasting Music Route.

Recorded Announcement Broadcast

The Recorded Announcement (RAN) Broadcast feature is applicable to RAN only, and the Music Broadcast feature is applicable to Music only.

Feature packaging

Music Broadcast (MUSBRD) is package 328. The following packages are also required to provide Music Broadcast capability:

- Music (MUS) package 44
- Recorded Announcement (RAN) package 7

To provide Music Broadcast capability to Enhanced Music (EMUS) features, the Enhanced Music (EMUS) package 119 is also required.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Change an existing non-broadcasting Music route to a broadcasting Music route.

2 LD 16 – Enable Conference-based Music route.

3 LD 14 – Configure Conference-based Music trunks.

Music Broadcast

LD 16 – Change an existing non-broadcasting Music route to a broadcasting Music route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	MUS	Music Trunk Data Block.
ICOG	OGT	Outgoing only Trunk.
...		
BDCT	YES	Allow Broadcast capability. NO = Deny Broadcast capability (default). If BDCT = YES, no conference loop is required. Each Music trunk has 64 broadcast connections.

Conference-based Music

LD 16 – Enable Conference-based Music route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	MUS	Music Trunk Data Block.

ICOG	OGT	Outgoing only Trunk.
...		
BDCT	NO	Deny Broadcast capability. YES = Allow Broadcast capability. If BDCT = YES, no conference loop is required. Each Music trunk has 64 broadcast connections.

LD 14 – Configure Conference-based Music trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	MUS	Music trunk.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
RTMB	0-511 1-510 0-127 1-510	Route number and Member number For Small Systems For Large Systems and Succession 1000 systems
...		
CFLP	0 - 158	Music Conference Loop. Prompted only for non-broadcasting Music routes.

Feature operation

No specific operating procedures are required to use this feature.

Meridian 1, Succession 1000,
Succession 1000M

Features and Services

Book 2 of 3 (D to M)

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NORTEL NETWORKS™

N Digit DNIS

Contents

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Feature description

Dialed Number Identification Services (DNIS) presents the Automatic Call Distribution (ACD) call to an agent's set or terminal. The incoming call displays the DNIS digits which represent product lines or services. The displayed DNIS digits reduces the time needed to service a call and the additional information helps the agent provide a greater degree of customer service. The ACD Routing by DNIS number routes the call to a specific ACD DN based on the DNIS number dialed.

With the N Digit DNIS feature, the DNIS length is 31 digits. Both ACD and Network ACD (NACD) support the N Digit DNIS feature.

Operating parameters

If the system initializes during an active call, DNIS information is lost.

M911 trunks cannot be configured as DNIS trunks.

System messages for the Time Slot Monitor (TSM) supports 31 digits of DNIS.

Applications and features display DNIS in the following ways:

- Meridian MAX 9.0 supports up to nine digits of DNIS information. Nine digits of DNIS information are sent over the High Speed Link (HSL). The first or last nine digits of DNIS information is sent depending on the configuration of the WDGT prompt in the RDB block.
- Auxiliary Processor Link (APL) supports four DNIS digits. If the DNIS information is longer than four digits, the first or last four digits are sent over the APL depending on the configuration of the WDGT prompt in the RDB block.
- Call Detail Recording (CDR) supports up to seven digits of DNIS digits. If more than seven digits of DNIS are received, the first or last seven digits are displayed on the CDR, depending on the configuration of the WDGT prompt in the RDB block.
- Call Party Name Display (CPND) supports name configuration up to seven digits of DNIS. If the DNIS information is more than seven digits, a name is not configured.
- Feature Group D supports seven digits of DNIS information.
- The agent's set is limited to 12 digits of DNIS display. If more than 12 digits of DNIS are received, the first 12 or the last 12 digits of DNIS are displayed, depending on the configuration of the WDGT prompt in the RDB.

Feature interactions

Automatic Call Distribution DNIS routing through IDC table

The Incoming Digit Conversion (IDC) table converts the DNIS digits to a valid DN. With the N Digit DNIS feature, the DNIS information is expanded to a range of one to31 digits. The maximum number of DNIS digits that are translated by the IDC tree to an internal DN is limited to 16, due to the DC feature.

Application Module Base

The system is connected to Application Module Base (AM Base) through Application Module Link (AML). DNIS information is in AML messages; therefore, the AM Base supports the expanded DNIS information.

Application Module Link (AML) messages

Call presentation and call modification receives DNIS through AML messages. Messages related to DNIS go through the AML to the Meridian Link Module to the Customer Controlled Routing (CCR).

Call Detail Recording

The Call Detail Recording supports up to seven DNIS digits. If the DNIS digits exceeds seven digits, the Call Detail Recorder (CDR) uses the first or last seven digits, depending on the configuration of the WDGT prompt in the RDB block.

Customer Controlled Routing

Customer controlled routing (CCR) uses the DNIS number to determine which call processing treatment is used for a DNIS trunk call.

Digit display for DNIS

The agent set is limited to a display of 12 DNIS digits. If the digits exceed the set's display capabilities, the first or last 12 DNIS digits are displayed depending on the configuration of the WDGT prompt in the RDB block.

Host Enhanced Routing

The Meridian Link's Host Enhanced Routing allows an incoming call to be routed before call termination. An Incoming Call (ICC) message sent to the Meridian Link Module contains calling party information, DNIS information, and Controlled Directory Number (CDN).

Meridian Link Interactions

Any ringing message sent to the Meridian Link over the AML contains expanded DNIS information. The Meridian Link sends this expanded information to the host application.

Meridian Mail

Meridian Mail receives DNIS digits over the Command Status Link (CSL). The DNIS message contains one to 31 DNIS digits, instead of the previously supported one to seven digits. Since Meridian Mail limits DNIS digits to 30, the AML message uses 30 digits.

Meridian MAX

The system communicates with Meridian MAX, ACD MAX, or ACD supports nine digits of DNIS.

Multi-Frequency Signaling for KD3 for Spain.

If a DNIS route uses Multi-Frequency Compelled (MFC) signals, the DNIS route must use the same number of digits as the MFC.

Multi-Frequency Signaling for Socotel

Multi-Frequency signaling for Socotel (MFE) trunks use either four or five signals, which requires DNIS to use the same number.

Network Automatic Call Distribution

The Network Automatic Call Distribution (NACD) sends and receives DNIS calls to a remote node through an NACD-Call Setup message. The remote node receives and saves the expanded one to 31digits of a DNIS message.

Symposium Call Center Server

The interaction of N Digit DNIS with Symposium Call Center Server (SCCS) is the same as its interaction with Customer Controlled Routing (CCR) and AM Base. Any AML message sent to the AM Base contains expanded DNIS information. AM Base supports the expanded DNIS information. Symposium supports seven digits of N Digit DNIS information.

Feature packaging

This feature is packaged as part of the existing DNIS package 98.

Feature packages required for the N Digit DNIS are:

- Dialed Number Identification System (DNIS) package 98

- Automatic Call Distribution (ACD A) package 45
- Digit Display (DDSP) package 19
- Incoming DID Digit Conversion (IDC) package 113
- New Format Call Detail Recording (FCDR) package 234
- New Flexible Code Restriction (NFCR) package 49

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Define SDI port for Auxiliary Processor Link.

2 LD 49 – Define Incoming Digit Conversion (IDC) table.

3 LD 16 – Define Incoming DID Digit Conversion DNIS route.

4 LD 14 – Define a trunk that auto-terminates on ACD-DNIS.

5 LD 16 – Define a route with DNIS feature enabled and AUTO-terminate.

6 LD 15 – Define APL Link number, enable the Incoming Digit Conversion (IDC) operation to include DNIS for a customer.

7 LD 23 – Define ACD group.

There are two configurations possible:

1 Define SDI port for Auxiliary Processor Link in LD 17.

2 Define Incoming Digit Conversion table in LD 49.

3 Define IDC-DNIS route in LD 16.

4 Define a trunk that auto-terminates on ACD-DNIS in LD 14.

OR

1 Define a route Auto Terminate Route in LD 16.

2 Define APL Link number, enable the Incoming Digit Conversion (IDC) operation to include DNIS for a customer in LD 15.

3 Define ACD group in LD 23.

LD 17 – Define SDI port for Auxiliary Processor Link.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	NEW TTY 0-15	Add an APL port.
CTYP	aaaa	Card type. aaaa = DCHI, SDI, SDI2, SDI4.
USER	APL	APL port connects to data link.

LD 49 – Define Incoming Digit Conversion (IDC) table.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	IDC	Type of data block (FCR or IDC).
CUST	xx	Customer number, as defined in LD 15
DCNO	0-254	Incoming Data Conversion (IDL) tree number.
...	...	...
...	...	...

LD 49 – Define Incoming Digit Conversion (IDC) table.

Prompt	Response	Description
IDGT	0-99999999 0-99999999	Incoming digits to be converted to ACD DN.
	<CR>	Re-prompt request.

LD 16 – Define Incoming DID Digit Conversion DNIS route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
...	...	...
AUTO	 NO	Auto-terminate. YES = The route members terminate on DN defined by response to ATDN prompt in LD 14. NO = The route members terminate normally.
DNIS	YES	ACD DNIS route.
--NDGT	1-(4)-31	Number of DNIS digits required on the route. The extension 31 digits is available only for DID, TIE or IDA routes.

--WDGT	(L)F	First or last DNIS digits to be sent on APL and HSL link. Where: F = First, L = Last (default) WDGT has no effect on AML Links. All DNIS digits are sent for AML. Prompted if NDGT is greater than four. Also used for CDR when the New Format CDR (FCDR) package 234 is disabled. First or last 4 digits for APL. First or last 12 DNIS digits for digit display. First or last 9 DNIS digits for MAX. First or last 7 DNIS digits for CDR.
--IDC	YES (NO)	Incoming DED digit conversion on this route YES = Allow Incoming DID Digit Conversion on this route. (NO) = Deny Incoming DID Digit Conversion on this route.
--DCNO	0-254	IDC translation table for this route in the day mode.
--NDNO	0-254	IDC Conversion Table for the night mode.

LD 14 – Define a trunk that auto-terminates on ACD-DNIS.

Prompt	Response	Description
REQ	NEW	Add a trunk.
TYPE	DID	Direct Inward Dialing trunk type.
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
ATDN	xxxx	xxxx = ACD-DN defined in LD 23.
CLS	DTN	Digitone signaling.

OR

LD 16 – Define a route with DNIS feature enabled and AUTO-terminate.

Prompt	Response	Description
REQ	NEW	Add a new data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
...	...	
AUTO	YES	Auto-terminate trunk. YES =YES = the route members terminate on DN defined by response to Auto Terminate Directory Number prompt in LD 14. (NO) =The route members terminate normally at the console.
DNIS	YES	ACD-DNIS route. YES = allow the ACD DNIS route. (NO) = Deny the ACD DNIS route. Prompted with Automatic Call Distribution Package D. (ACCDD) package 50, and the RTYP = TIE or Direct Inward Dialing (DID).
NDGT	1-(4)-7 1-(4)-31	Number of DNIS digits required on the route. The extension to 31 digits is available only for DID, TIE or IDA routes.
WDGT	(L) F	First or last 4 DNIS digits to be sent on APL and HSL link. WDGT has no effect on AML links. All DNIS digits are sent for AML. Prompted if NDGTR is greater than 4. Also used for CDR when the New Format CDR (FCDR) package 234 is disabled. ***Note:*** The number of (MFX), MFE or MFC digits takes precedence over the number of DNIS digits that are configured.

LD 15 – Define APL Link number, enable the Incoming Digit Conversion (IDC) operation to include DNIS for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:		Disable/Enable New flexible code Restriction.
	FCR	Flexible Code Restriction.
CUST		Customer number
	0-99	For Large Systems
	0-31	For Small Systems and Succession 1000 systems
NFCR		New Flexible Code Restriction. (NO) = Default, disable New Flexible Code Restriction.
	YES	YES = Enable New Flexible Code Restriction. To build an Incoming Digit Conversion (IDC) table in LD 49, NFCR and Incoming DID Digit Conversion (IDCA) must be set to YES. NFCR is prompted with New Flexible Code Restriction (NFCD) package 49.
-MAXT	1-255	Maximum number of New Flexible Code Restriction (NFCR) tables. Once defined a lower value cannot be entered for MAXT. The sum of the values for MAXT + DCMX $\leq$ 255 per customer.
IDCA		Incoming DID Digit Conversion.
	YES	(NO) = Default. Deny Incoming DID Digit Conversion. YES = Allow Incoming DID Digit Conversion. NFCR must = YES before IDCA can = YES. Prompted with Incoming Digit Conversion (IDL) package 113.
-DCMS	1-255	Digit conversion maximum number of tables (DCMS). The sum of the values for MAXT and DCMX cannot exceed 255 or MAXT + DCMX = 255.

LD 23 – Define ACD group.

Prompt	Response	Description
REQ	NEW	Add ACD group.
TYPE	ACD	ACD data block.
CUST	xx	Customer number, as defined in LD 15
ACDN	xxxx	ACD Directory Number.

Feature operation

No specific operating procedures are required to use this feature.

New Flexible Code Restriction

Contents

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Feature description

New Flexible Code Restriction (NFCR) controls the access of Toll Denied terminals to outgoing trunk routes and digits dialed on them. Calls are allowed or denied based on the specific digit sequence dialed.

Toll Denied (TLD, CTD, CUN) telephones and trunks are assigned a Network Class of Service (NCOS) and are allowed or denied calling privileges according to the Facility Restriction Level (FRL) assigned to their NCOS. If, however, a user who has CTD or CUN Class of Service has dialed the call using a Basic Alternate Route Selection (BARS), Network Alternate Route Selection (NARS), Coordinated Dialing Plan (CDP), or Automatic Number Identification (ANI) access code, the NFCR restrictions do not apply. For these users, NFCR applies only on direct trunk access code type calls. TLD users are always affected no matter how their call is dialed.

When a user accesses an outgoing route, the user's assigned FRL determines which digits are allowed or denied on that route. Up to eight FRL codes can be assigned per trunk route. When a user dials denied digits following direct trunk access codes, intercept treatment is given. NFCR can be programmed to deny certain outpulsed digits, not dialed digits, when Electronic Switched Network (ESN) calls are to be denied for TLD users.

Using "code restriction trees," NFCR can be programmed to analyze each digit individually and allow or deny a call on the basis of any digit or digit sequence dialed. There can be up to 255 code restriction trees per customer group. Each trunk route can access up to eight trees, and each tree can be used by more than one route. The code restriction tree corresponding to the terminal user's FRL is defined by the trunk route. Digits can also be bypassed and allowed to process with no restriction; however, certain digits that follow these might be restricted.

NFCR can be programmed to count the number of digits dialed and deny any call exceeding the specified number of digits. If a user dials an octothorpe (#) before NFCR has finished digit counting, the call is disallowed and intercept treatment is given. This prevents digits from 2500 sets or Dual-tone Multifrequency (DTMF) trunks from being outpulsed before being counted or analyzed by code restriction. Up to 50 digits can be analyzed.

Operating parameters

New Flexible Code Restriction (NFCR) can be programmed to count the number of digits dialed and deny any call exceeding the specified number of digits.

Only the digits zero (0) through nine (9) are considered. If a user dials an asterisk (*), it is not counted as a dialed digit. If the user dials an octothorpe (#) before NFCR has finished digit counting, the call is disallowed and the appropriate intercept treatment is provided. This prevents digits from 2500-type telephones or Dual-tone Multifrequency (DTMF) trunks from being outpulsed before being counted or analyzed by code restriction.

As many as 255 code restriction trees are available per customer. Eight code restriction trees can be referenced by each trunk route.

Up to 50 digits can be analyzed by NFCR.

When Code Restriction (LD 19) and NFCR (LD 49) are both enabled for the same customer, NFCR takes precedence. Any parameters required for Code Restriction are ignored.

Feature interactions

Access Restrictions

The Code Restriction feature and New Flexible Code Restriction cannot be implemented simultaneously for the same customer.

Attendant Blocking of Directory Number

When the attendant has a blocked DN on the source side and dials on the destination side, any New Flexible Code Restriction active for the set of the blocked DN will be overridden. This is the same as if the attendant had a normal established call to the DN on the source side and dials the destination side.

Authorization Code Security Enhancement

If the Class of Service of the authorization code is Toll Denied (TLD), NFCR is applied. If the Class of Service is Conditionally Unrestricted (CUN) or Conditionally Toll Denied (CTD) and the call is not routed through BARS/ NARS, CDP or ANI, NFCR is applied.

Automatic Number Identification

Calls from Toll Denied (TLD) stations routed by Automatic Number Identification (ANI) are subject to NFCR. Calls placed by Conditionally Toll Denied (CTD) and Conditionally Unrestricted (CUN) Class of Service stations subject to ANI are treated as unrestricted calls.

Automatic Redial

Automatic Redial (ARDL) calls must pass New Flexible Code Restriction (NFCR) checks. If the redialed number is restricted, the ARDL request is cancelled.

Basic Alternate Route Selection (BARS)
Network Alternate Route Selection (NARS)
Coordinated Dialing Plan (CDP)

Only TLD telephones are subject to NFCR when calls are routed by BARS/NARS/CDP. CTD and CUN calls routed by BARS/NARS/CDP are not subject to NFCR treatment.

China – Flexible Feature Codes - Outgoing Call Barring
Enhanced Flexible Feature Codes - Outgoing Call Barring

Outgoing Call Barring uses NFCR trees to define the digit sequences that are not allowed for each level of barring. However, OCB analyses all dialed digits, whereas NFCR only analyses digits outpulsed on trunks. This means that the same tree will not normally be usable for both features, unless only Coordinated Dialing Plan trunk calls are to be blocked for both features and no digit manipulation is done.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

Toll-denied users (CLS = TLD) may be subject to NFCR if they make a NARS call across the DPNSS1 UDP network. The New Flexible Code Restriction feature is supported in a DPNSS1 UDP network.

Direct Inward System Access

If the Direct Inward System Access (DISA) DN has a TLD, CUN, or CTD Class of Service, calls made through DISA are eligible for NFCR treatment.

Electronic Lock Network Wide/Electronic Lock on Private Lines

With NFCR, toll denied stations are allowed or denied calling privileges according to the Facility Restriction Level (FRL) assigned to the NCOS defined in the protected line block. For a locked set, NCFR uses the FRL assigned to the CNCS to determine its calling privileges if one is defined; if no CNCS is defined, the NCOS of the locked set will be used.

Federal Communications Commission Compliance for Equal Access

The New Flexible Code Restriction (NFCR) feature has been modified to allow for the restriction of Equal Access international toll calls (10XXX+011+CC+NN) while not restricting Equal Access operator calls (10XXX+0).

Forced Charge Account

Calls placed through the Forced Charge Account feature are not eligible for NFCR treatment.

Network Class of Service

Toll Denied stations and trunks must have a Network Class of Service (NCOS) assigned to be allowed or denied calling privileges by NFCR. This is because the FRL associated with the NCOS of the user determines which codes are allowed or denied on an outgoing trunk call. The range of NCOS groups varies as follows:

(0)-3 for standalone CDP
(0)-7 for BARS/CDP and NFCR
(0)-15 for NARS and NFCR
(0)-99 for BARS/NARS/CDP/NFCR

Scheduled Access Restrictions

Associating an FRL with a different NFCR tree affects any Network Class of Service (NCOS) that uses that FRL. Each such NCOS assigned to a Scheduled Access Restrictions (SAR) group might need to be reconsidered. Also, different facility restriction levels and NFCR trees are used at different times according to the NCOS assigned to the SAR group.

Feature packaging

New Flexible Code Restriction (NFCR) package 49 requires:

- Network Class of Service (NCOS) package 32.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable NFCR for a customer.

2 LD 87 – Define NCOS groups and associated FRL.

3 LD 49 – Add, change, or print code restriction trees.

4 LD 16 – Associate an FRL with a code restriction tree.

5 LD 10 – Assign an analog (500/2500 type) telephone a Toll Denied and Network Class of Service.

6 LD 11 – Assign Meridian 1 proprietary telephones a Toll Denied and Network Class of Service.

7 LD 14 – Assign a trunk a Toll Denied and Network Class of Service.

8 LD 24 – Assign a DISA data block a Toll Denied and NCOS.

9 LD 88 – Assign an Authorization code a Toll Denied and NCOS.

LD 15 – Enable NFCR for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FCR	New Flexible Code Restriction options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
NFCR	(NO) YES	(Disable) enable NFCR.
- MAXT	1-255	Maximum number of code restriction trees.

LD 87 – Define NCOS groups and associated FRL.

Prompt	Response	Description
REQ	NEW CHG	Create new, or change.
CUST	xx	Customer number, as defined in LD 15
FEAT	NCTL	Network Control.
NCOS	(0)-99	NCOS group.
FRL	0-7	FRL is assigned to each NCOS. It determines the entries in a route list (RLI) to which it has access. 0 is the most restrictive, 7 is the least restrictive and can access more entries.

LD 49 – Add, change, or print code restriction trees.

Prompt	Response	Description
REQ	NEW CHG PRT	Create new, change, or print data.
TYPE	FCR	NFCR data block.
CUST	xx	Customer number, as defined in LD 15
CRNO	(0)-254	Code restriction tree number.
INIT	ALOW DENY	Allow or deny all codes.
The following prompts appear if INIT = ALOW		
DENY	xx...xx	Digit sequence to be denied.
ALOW	xx...xx	Digit sequence to be allowed.
BYPS	xx...xx	Digit sequence to be bypassed.
The following prompts appear if INIT = DENY		
ALOW	xx...xx	Digit sequence to be allowed.

DENY	xx...xx	Digit sequence to be denied.
BYPS	xx...xx	Digit sequence to be bypassed

LD 16 – Associate an FRL with a code restriction tree.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
FRL	x yyy	x = FRL number (0-7). yyy = code restriction tree number (1-255). FRL is reprompted to allow input of eight FRLs. A carriage return causes the next prompt to appear.

LD 10 – Assign an analog (500/2500 type) telephone a Toll Denied and Network Class of Service.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
NCOS	(0)-99	NCOS.
CLS	TLD	Toll Denied Class of Service.

LD 11 – Assign Meridian 1 proprietary telephones a Toll Denied and Network Class of Service.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
NCOS	(0)-99	NCOS.
CLS	TLD	Toll Denied Class of Service.

LD 14 – Assign a trunk a Toll Denied and Network Class of Service.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where: aaa = CSA, TIE, or WAT.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
NCOS	(0)-99	NCOS.
CLS	TLD	Toll Denied Class of Service.

LD 24 – Assign a DISA data block a Toll Denied and NCOS.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	DIS	DISA data block.

CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Security password.
DN	xxx....x	DISA Directory number.
NCOS	(0)-99	NCOS.
COS	TLD	Toll Denied Class of Service.

LD 88 – Assign an Authorization code a Toll Denied and NCOS.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	AUB	Authorization code data block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Security password.
CLAS	(0)-115	Class code to be assigned.
NCOS	(0)-99	NCOS.
COS	TLD	Toll Denied Class of Service.

Feature operation

No specific operating procedures are required to use this feature.

Night Key for Direct Inward Dialing (DID) Digit Manipulation

Contents

This section contains information on the following topics:

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Feature description

The Night Key for DID Digit Manipulation (NKDM) uses DID Incoming Digit Conversion (IDC) to convert received DID digits into a Night Service Directory Number (DN). NKDM is used to switch between Night and Day modes.

The Day/Night mode is controlled by a DID Route Control (DRC) key on an Attendant Console, or Meridian 1 proprietary telephone. There can only be one DRC key for each DID route.

The Night tree table is invoked in any of the following ways:

- when the attendant goes into Night Service, or the last attendant activates the POS BUSY key (provided that Attendant Overflow Position is not equipped)

- when an attendant activates the DID Route Control (DRC) key
- when a Console Presentation Group (CPD) attendant goes into Night Service, or
- when a Meridian 1 proprietary telephone activates the DRC key.

In each case, only the DID routes controlled by the initiating source (console or telephone) are affected.

Operating parameters

The maximum number of conversion tables per customer is 255. These tables are shared between the Incoming Digit Conversion (IDC) and the New Flexible Code Restriction (NFCR) trees.

When an attendant activates the DID Route Control Key (DRC), the M2250 Digital Attendant Console going into Day/Night mode does not change the Incoming Digit Conversion (IDC) table that is used. The DRC takes precedence over the M2250 Console and ignores the M2250 state if a DRC is assigned.

> ***Note:*** The DRC key can only be configured on keys with lamp indicators.

For each DID route, there is only one configured DRC key per telephone.

When using the Night tree table, the same assumptions that apply to Incoming Digit Conversion (IDC) apply to this feature. The Night tree table for DID Digit Manipulation (NKDM) applies only to DID routes.

For a Dialed Number Identification Service (DNIS) route, make sure that the correct table is selected for the conversion of incoming digits.

Feature interactions

Attendant Administration

The DID Route Control (DRC) key is not supported by Attendant Administration.

Attendant Overflow Position

When the last attendant activates the POS BUSY key, the system does not go into Night Service if an Attendant Overflow Position Directory Number (DN) is available.

Automatic Set Relocation

Delete the DRC key from a telephone before performing Automatic Set Relocation. If this is not done, the DRC lamp is activated on the wrong telephone.

Console Presentation Group Level Services

The Day/Night table can be activated with the DRC key by any attendant in the Console Presentation Group (CGP).

Feature packaging

The Night Key for DID Digit Manipulation (NKDM) is part of base system software. The following packages are required:

- Network Class of Service (NCOS) package 32
- New Flexible Code Restriction (NFCR) package 49, and
- Incoming Digit Conversion (IDC) package 113.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Incoming Digit Conversion for Night mode.

2 LD 49 – Add, change, or print code restriction trees.

3 LD 16 – Set IDC tree for Night mode. Note that a DID route cannot be removed if it is controlled by a DCR key.

4 LD 12 – Define a DID Route Control (DRC) key on an Attendant Console.

5 LD 11 – Define a DRC key on a Meridian 1 proprietary telephone.

LD 15 – Enable Incoming Digit Conversion for Night mode.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CDB FCR	Customer Data Block. New Flexible Code Restriction options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
NFCR	(NO) YES	Enable New Flexible Code Restriction.
- MAXT	1-255	Maximum number of NFCR trees.
IDCA	(NO) YES	Enable IDC. IDC cannot be disabled if any telephone has a DCR key.
- DCMX	1-255	Maximum number of IDC conversion tables. The sum of the values of MAXT and DCMX cannot exceed 255 per customer.

LD 49 – Add, change, or print code restriction trees.

Prompt	Response	Description
REQ	NEW CHG PRT	Create new, change, or print data.
TYPE	IDC	NFCR data block.
CUST	xx	Customer number, as defined in LD 15

DCNO	0-254	IDC tree number.
IDGT	0-9999 0-9999	Directory Number (DN) or range of DNs to be converted. The external DN to be converted is output and the user enters the internal DN. For example, to convert the external DN 3440 to 510, enter 3440. The system prompts 3440 and you enter 510.

LD 16 – Set IDC tree for Night mode. Note that a DID route cannot be removed if it is controlled by a DCR key.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
TKTP	DID	DID route.
IDC	(NO) YES	Enable IDC.
DCNO	0-254	IDC tree for Day mode.
NDNO	0-254 <CR>	IDC tree for Night mode. Set tree to the same number as Day mode (the default).

LD 12 – Define a DID Route Control (DRC) key on an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx DRC	DID Route Control key, where: xx = key number 0-9 (0-19 on the M2250).

LD 11 – Define a DRC key on a Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx DRC yyy	DRC, where: xx = key number, and yyy = route number (0-511).

Feature operation

Follow these steps to change one DID route to Day/Night mode from the Attendant Console:

1 Select an idle loop key.

2 Press **DRC** and dial the access code of the DID route (ACOD).

If the DRC indicator is on steadily, the route is in Day mode.

If the DRC indicator is flashing, the route is in Night mode.

3 Press **DRC** again.

If the DRC indicator was on steadily, the route is put into Night mode.

If the DRC indicator was flashing, the route is put into Day mode.

Follow these steps to change all DID routes to Day/Night mode from the Attendant Console:

1 Select an idle loop key.

2 Press **DRC** and dial the octothorpe (#).

If the DRC indicator is on steadily, all routes are in Day mode.

If the DRC indicator is flashing, one or more routes are in Night mode.

3 Press **DRC** again.

If the DRC indicator was on steadily, all routes are put into Night mode.

If the DRC indicator was flashing, all routes are put into Day mode.

Note: To change from some routes in Night mode to all routes in Night mode, you must first put all routes into Day mode.

Follow these steps to change one DID route to Day/Night mode from a telephone:

1 Check the **DRC** indicator.

If the DRC indicator is on steadily, the route is in Day mode.

If the DRC indicator is flashing, the route is in Night mode.

2 Press **DRC**.

The route toggles between Night and Day mode.

Night Restriction Classes of Service

Contents

This section contains information on the following topics:

Feature description 99
Operating parameters 99
Feature interactions 100
Feature packaging 100
Feature implementation 101
Feature operation 103

Feature description

The purpose of the Night Restriction Classes of Service (NRCLS) feature is to restrict the operation of the Call Waiting, Forced Camp-on, and Priority Override features so they operate during Night Service only. Therefore, the NRCLS feature will apply to any set which has Call Waiting, Forced Camp-on, or Priority Override features equipped.

Operating parameters

The Night Restriction Classes of Service (NRCLS) feature is available on any station.

Feature interactions

Call Waiting

If Call Waiting and Night Restriction for Call Waiting Class of Service (NRWA) are assigned, Call Waiting will be operational for the set only when Night Service is in effect.

Call Waiting Redirection

The Call Waiting Redirection feature applies to unanswered calls given Call Waiting treatment when the Night Restriction Classes of Service feature allows Call Waiting.

Camp-on, Forced

If Forced Camp-on and Night Restriction for Forced Camp-on Class of Service (NRCA) are assigned, Forced Camp-on will be operational for the set only when Night Service is in effect.

Override

If Priority Override and Night Restriction for Priority Override Class of Service (NROA) are assigned, Priority Override will be operational for the set only when Night Service is in effect.

Feature packaging

The Night Restriction Classes of Service feature is packaged under the Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 and LD 11 – These overlays are modified to accept the following six new classes of service: NRCD, NRCA, NROD, NROA, and NRWD, NRWA.

2 LD 81 – This overlay prints DES to TN and last service change information for selected features. The classes of service NRCA, NRCD, NROA, NROD, NRWA, and NRWD are now allowed.

LD 10 and LD 11 – These overlays are modified to accept the following six new classes of service: NRCD, NRCA, NROD, NROA, and NRWD, NRWA.

Prompt	Response	Description
REQ:	CHG NEW	Change, or add.
...		
CLS		Class of Service.
	(NRCD) NRCA	Night Restriction of forced Camp-on (Denied) Allowed. Forced Camp-on must be configured for the set. Assigning NRCD Class of Service allows Forced Camp-on to operate during both Night and Day Service. Assigning NRCA Class of Service restricts Forced Camp-on to operate during Night Service only. Default is NRCD.

	(NROD) NROA	Night Restriction of priority Override (Denied) Allowed. Priority Override must be configured for the set. Assigning NROD Class of Service allows Priority Override to operate during both Night and Day Service. Assigning NROA Class of Service restricts Priority Override to operate during Night Service only. Default is NROD.
	(NRWD) NRWA	Night Restriction of call Waiting (Denied) Allowed. Call Waiting must be configured for the set. Assigning NRWD Class of Service allows Call Waiting to operate during both Night and Day Service. Assigning NRWA Class of Service restricts Call Waiting to operate during Night Service only. Default is NRWD.

LD 81 – This overlay prints DES to TN and last service change information for selected features. The classes of service NRCA, NRCD, NROA, NROD, NRWA, and NRWD are now allowed.

Prompt	Response	Description
REQ	LST	List telephones equipped with the feature specified by the prompt FEAT.
	CNT	Print a count of telephones equipped with the feature specified by the prompt FEAT.
	END	End overlay activity.
CUST	xx	Customer number, as defined in LD 15
DATE	1-31 Jan-Dec	Print data from activity date specified.
	ACT	Print data from last activity date.
	<CR>	Disregard date restrictions.
PAGE	(NO) YES	Print data on a per page basis.

DES	XXXXXX X+ + <CR>	Print station with designator XXXXXX. Print data for stations with designators starting X. Print data for all stations with no designator. Print data for all stations with designators.
FEAT	NRCA NRCD NROA NROD NRWA NRWD	Night Restriction of Forced Camp-on Allowed, or Denied. Night Restriction of Priority Override Allowed, 0r Denied. Night Restriction of Call Waiting Allowed, or Denied.

Feature operation

A customer or a Console Presentation Group (CPG) can be put into Night Service manually by pressing the Night key on the Attendant Console or automatically by Scheduled Access Restriction (SAR) or Attendant Forward No Answer (AFNA).

Depending on the Class of Service (CLS) and key assignments, the operation of the features will be allowed or denied as summarized in the following table:

Figure 1
Feature operation summary

CLS	Feature X Allowed	Feature X Denied
NRXA	Feature **X** is restricted to operate during Night Service only.	Feature **X** always denied.
NRXD	Feature **X** operates whether Night Service is active or not.	Feature **X** always denied.

Legend:
NR**X**A: Night Restriction of feature **X** Allowed for this set.
NR**X**D: Night Restriction of feature **X** Denied for this set.

Where **X** =:
W for Call Waiting
C for Forced Camp-on, or
O for Priority Override.

Night Service

Contents

This section contains information on the following topics:

Feature description 105
Operating parameters 106
Feature interactions 107
Feature packaging 112
Feature implementation 112
Feature operation 113

Feature description

Night Service permits incoming calls normally directed to the attendant to be routed to a defined destination. A separate Night key/lamp pair allows the attendant to put the system into Night Service.

Three types of Night Service are provided which the customer can specify separately or in any combination:

- Selected Trunks to Selected Directory Number (DNs): Some or all of the trunks can be assigned to ring selected DNs when the system is in Night Service. The assignment of trunks to stations can be modified by the attendant or by a service change.

- Night Answer Telephone: All calls normally routed to the Attendant Console can be routed to one particular DN that is designated as the night answer destination for the customer. Trunk Answer From Any Station (TAFAS) can be used to pick up calls routed to this number. With TAFAS in effect, incoming calls activate a common alerting device, such as a bell, when the system is in Night Service. Any user can answer the call by dialing the Special Prefix (SPRE) code and then pressing 4.
- Night Service by Time of Day (NSTD): NSTD allows one of a group of Directory Numbers (DNs) to be selected for call routing based on the time of day instead of all calls being routed to a fixed Night Service DN. NSTD allows the definition of up to four Night DNs with a time associated with each. Calls are forwarded to the appropriate DN by the associated time.

Operating parameters

Night Service can only be activated from the Attendant Console.

Any restrictions or features assigned to the night answering station apply. Therefore, a fully restricted (FRE) Class of Service should not be used for Night Service Directory Numbers (DNs), unless the FRPT prompt in LD 17 is OLFR (allow FRE telephones to serve as a Night DN).

A bell circuit or alerting device must be provided by the customer for TAFAS. This device must be compatible with the 20 Hz ringing signal (that is, two seconds on, four seconds off).

If a trunk is assigned a Night DN other than the Night Answer Number defined in the Customer Data Block, incoming calls to that trunk cannot be picked up with the TAFAS feature. Assignment in LD 14 takes precedence over the Customer Data Block.

If an attendant is not assigned to a customer, the customer is automatically in Night Service upon system start-up. The following tables show how calls are directed during Night Service, depending on the time of day:

Call is directed to Night DN	Between times:
NIT1 DN	TIM1 and TIM2
NIT2 DN	TIM2 and TIM3
NIT3 DN	TIM3 and TIM4
NIT4 DN	TIM4 and TIM1

It is possible to remove a defined night DN without modifying the other DNs. For example, if NIT3 is removed, calls are directed as follows:

Call is directed to Night DN	Between times:
NIT1 DN	TIM1 and TIM2
NIT2 DN	TIM2 and TIM4
NIT4 DN	TIM4 and TIM1

Feature interactions

Attendant Overflow Position

A call rerouted through the Attendant Overflow Position feature is not redirected to the Night DN if the system is subsequently put into Night Service. When all Attendant Consoles are in Position Busy, the system will not go into Night Service until the AOP Busy key is activated.

Deactivating the AOP Busy key after the system has been placed in Night Service does not affect the Night Service feature.

Attendant Position Busy

When the last console operator activates the Position Busy key or the Night key, Night Service is put into effect. Incoming calls receive the customer-specified night treatment.

Automatic Wake Up

Unanswered Automatic Wake Up calls going through Attendant Recall are discarded if the Attendant Console is in the Night Service mode. Automatic Wake Up may still be programmed when the Attendant Console is in Night Service.

Call Forward Busy

When the system is in Night Service, Direct Inward Dialing calls forwarded by Call Forward Busy are routed to the specified night number. If the night telephone is busy, subsequent calls receive busy tone.

Call Pickup Network Wide

The Call Pickup Network Wide feature can be used to pick up a call to the night number if it is ringing an ordinary station (that is, analog (500/2500 type) telephone, 16-button Dual-tone Multifrequency, or proprietary set).

Call Waiting Redirection

Night Service has the same interaction with the Call Waiting Redirection feature as attendant-extended calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also has precedence over the Call Waiting recall timer.

Calls Waiting in Attendant Queue

Incoming calls ringing at the Attendant Console at time changeover are routed to the Night DN that just expired. New calls are routed to the new Night DN. If the attendant cancels Night Service, new calls are presented to the Attendant Console.

Once a call begins ringing at a Night DN, it stays there even if Night Service is cancelled or the timer expires.

Departmental Listed Directory Number

Departmental Listed Directory Number does not affect Night Service (including TAFAS). Calls presented to the LDN from an external source will queue for the night bell. All other attendant calls receive busy treatment if the night Directory Number (DN) is busy.

Directory Number Expansion

If the Directory Number Expansion (DNPX) package is equipped, the Night DNs can be up to seven digits; otherwise, the DN can be a maximum of four digits.

Distinctive/New Distinctive Ringing

Incoming calls terminating on a night Directory Number (DN) ring distinctively.

DPNSS1 Diversion

If a diverted call encounters an attendant in night service, the call receives Night Service Diversion if available.

End-to-End Signaling

Night Service works together with Attendant End-to-End Signaling (AEES). However, do not press this feature key while using AEES, or the Dual-tone Multifrequency (DTMF) code signals may be blocked.

Equi-distribution Network Attendant Service Routing

When the attendant goes into Night Service, calls presented to the attendant receive NAS routing in an attempt to reach another attendant that is in day service, rather than being routed to the local night DN.

Manual Line Service

When the system is in Night Service (NSVC) mode, all telephones with a manual Class of Service are routed to the telephone designated as the night number for the customer group.

Meridian 911 Call Abandon

Abandoned calls can be forwarded to the Night Call Forward DN if the Night Forward DN is an ACD DN. If a primary answering center goes into Night Service while there are abandoned calls in the queue, those abandoned calls are dropped. A CDR N record is printed if CDR is configured.

Multi-Party Operations

During Night Service, mishandled calls are routed to the night DN. External calls, other than DID calls, are queued until answered. TIE calls are disconnected if the night DN is busy.

Night Service

If the system is in Night Service mode, mishandled calls which are routed to the attendant are rerouted to the appropriate Night Service DN. External trunk calls, other than DID, are queued till they are answered.

TIE trunk calls are not queued at the Night Service DN. If the Night Service DN is busy, TIE calls are disconnected.

Night Service Enhancements

When the Night Service key is pressed on any Attendant Console, the customer enters Night Service and all Attendant Consoles are made Position Busy. It is then necessary to check all consoles for presented but unanswered calls, which must be cleared and requeued.

Recorded Overflow Announcement

The Recorded Overflow Announcement feature is inactive when the system is in Night Service.

Series Call

If the attendant extends a Series Call and goes into Night Service before it recalls to the attendant, the call recalls to the night DN and Series Call treatment is cancelled.

Trunk to Trunk Connection

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

Position Busy

When all attendants activate the Position Busy key, Night Service is in effect unless the Attendant Overflow Position (AOP) feature is equipped. If AOP is equipped, the Night key must be pressed to invoke Night Service. A call that is rerouted due to AOP is not redirected to the Night DN if the system is subsequently put into Night Service.

Night Service by Time of Day (NSTD) interactions

Call Park Recall

Calls parked by the attendant recall on the Night Service DN that is current at the time of recall.

Calls Waiting in Attendant Queue

Incoming calls ringing at the Attendant Console at time changeover are routed to the Night DN that just expired. New calls are routed to the new Night DN. If the attendant cancels Night Service, new calls are presented to the Attendant Console.

Once a call begins ringing at a Night DN, it stays there even if Night Service is cancelled or the timer expires.

Multi-Tenant Night Service

The same conditions that apply to the customer night number also apply to the Multi-Tenant Night Service. Console Presentation Group (CPG) allows separate night treatment for each tenant.

Meridian 911 Call Abandon

Abandoned calls are part of the transition mode when agents go to Night Service and the supervisor selects transition mode.

Series Call

If the attendant extends a Series Call and goes into Night Service before it recalls to the attendant, the call recalls to the night DN and Series call treatment is canceled.

Trunk Answer from Any Station

When a DN changeover occurs while an incoming call is ringing the current Night DN and a new incoming call is ringing the new Night DN, a user activating Trunk Answer from Any Station (TAFAS) picks up the call from the Night DN that just expired. However, if the ringing call is not picked up within one minute after the Night DN time changeover, the user can no longer pick up the call using TAFAS.

Trunk to Trunk Connection

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Add or change Night Service for a customer.

2 LD 14 – Add or change Night Service DN for trunks.

LD 15 – Add or change Night Service for a customer.

Prompt	Response	Description
REQ:	CHG	Change.

TYPE:	NIT	Night Service Options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- NIT1	xxx...x, X	Night Service DN 1 (enter X to remove). Night Service DN times must be defined in ascending order.
- TIM1	0-23 0-59	DN 1 time (hour and minute).
- NIT2	xxx...x, X	Night Service DN 2 (enter X to remove).
- TIM2	0-23 0-59	DN 2 time (hour and minute).
- NIT3	xxx...x, X	Night Service DN 3 (enter X to remove).
- TIM3	0-23 0-59	DN 3 time (hour and minute).
- NIT4	xxx...x, X	Night Service DN 4 (enter X to remove).
- TIM4	0-23 0-59	DN 4 time (hour and minute).

LD 14 – Add or change Night Service DN for trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	COT	Trunk type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
NITE	xxx...x, X	Night Service DN for this trunk (enter X to remove).

Feature operation

To place a customer into Night Service:

- Press **Shift** plus  at any console, or unplug all handsets and headsets.

To cancel Night Service when all handsets and headsets are unplugged:

- Plug in at least one handset or headset.

To cancel Night Service at a console when a handset or headset is plugged in:

- Press **Shift** plus 

  Note: If all Attendant Consoles are put in Position Busy, the system automatically goes into Night Service.

Night Service Enhancements

Contents

This section contains information on the following topics:

Feature description 115
Operating parameters 117
Feature interactions 119
Feature packaging 122
Feature implementation 123
Feature operation 124

Feature description

Night Service Enhancements introduces the following capabilities:

- All Calls Remain Queued for Night Service
- Recall to Night DN
- Requeuing of Attendant Presented Calls
- Camp-on from Inquiry Call (Station Camp-on)

All Calls Remain Queued for Night Service

This capability ensures that when Night Service is activated all calls in the attendant queue remain queued for Night Service treatment. Depending on the call type, the call may be presented to the Night DN, or continue waiting for the called party to answer. This includes Call Forward No Answer calls, recalls, and transfers to the attendant.

This capability applies to both standalone and networking environments. Within a networking environment, if Network Attendant Service (NAS) is equipped at all nodes, the calls are presented to a remote attendant, remote Night DN, or local Night DN, depending on the NAS configuration. This treatment applies to external calls only, since internal calls are not queued against a remote Night DN. If NAS routing is not involved, external calls are presented to the local Night DN.

Recall to Night DN

If the attendant camps-on party A to a busy set B, then goes into Night Service, the recall goes to the Night DN only if A is an external party (that is, CO, DID, FEX, WATS). This happens for a local camp-on.

For a Meridian Customer Defined Network (MCDN) camp-on with A at the far end of the MCDN NAS network and for a DPNSS1 camp-on with A at the far end of the DPNSS1 network the situation is as follows. If A is an internal party, the recall is left in the attendant queue, and can be answered by the attendant if the attendant returns to day service.

This internal/external difference does not hold true if the International Supplementary Features (SUPP) package 131 is equipped.

Requeuing of Attendant Presented Calls

The Requeuing of Attendant Presented Calls is an enhancement to the Attendant Forward No Answer feature. If a call presented to an Attendant Console is not answered, pressing the Position Busy key causes the call to be placed in the attendant queue.

If the console is the customer's last-active console, and Attendant Overflow Position (AOP) is active, a ringing call or a Call Waiting recall on the Destination side is disconnected. This ensures that any queued call will be presented at the AOP.

Any call presented at the AOP is not removed from the console and requeued if the Position Busy key is pressed.

The call is removed unanswered only if the Attendant Forward No Answer feature is active. In this case, after the Attendant Forward No Answer time out expires, the call is requeued and the AOP is idled.

All consoles will enter the Position Busy state if the Night Service key is pressed on any of the customer's consoles. Therefore, all consoles should be checked for presented, but unanswered calls, which have been requeued.

Camp-on from Inquiry Call (Station Camp-on)

With this feature, any internal station can camp an external call on to another internal station that is busy. Prior to the introduction of this feature attendant's were the only parties that could camp calls on to busy internal stations. The term internal station includes stations on other nodes within an Meridian Customer Defined Integrated Services Digital Network (MCDN).

When a transferring party reaches a busy desired internal party, the transferring telephone will receive ringback tone (providing certain conditions are met). When the transferring party completes the transfer, the external (calling) party will Camp-on to the desired party and the external party (an external party is any CO, DID, FEX, or WATS call) will receive ringback tone or announcement.

This feature applies to both standalone and network environments.

Within a network environment, the transferring and Camped-on to stations may be on the same or different nodes, as long as all nodes are configured with Network Station Camp-on.

Operating parameters

Camp-on from Inquiry Call (Station Camp-on)

The restrictions which currently apply to the operation of the Camp-on feature from an Attendant Console also apply to Camp-on from Inquiry Call (Station Camp-on).

These restrictions are:

- Camp-on will not be permitted if the desired station is in a state other than established (for instance, ringing or dialing).
- Only one call at a time may be Camp-on a busy station.
- Calls cannot Camp-on to a station with the Call Waiting feature configured.
- The station camped-on to will be given Warning Tone only if the customer has Camp-on Tone Allowed (CTA) in the Customer Data Block (LD 15) and the station has Warning Tone Allowed (WTA) Class of Service assigned. If the station has Warning Tone Denied (WTD) Class of Service assigned the Camp-on will take effect without giving any Camp-on Tone to the camped-on to (desired) party.
- The transferring station will receive Busy Tone only if the response to the STCB prompt in the Customer Data Block (LD 15) of the Camped-on to (desired) set is YES. Otherwise, the transferring station will receive ringback tone.

Camp-on Indication

When a call is extended from an attendant to a busy station there is a specific combination of tones and indicator states to identify the Camp-on state.

When an inquiry call is made from a station, there is only one way for the user to distinguish between a busy set and an idle ringing set. That way is to ensure that the response to the STCB prompt in the Customer Data Block (LD 15) of the Camped-on to (desired) set is YES. Otherwise, ringback tone is provided in both cases.

Night DN
Recall to Night Directory Number

When the customer goes into Night Service, if the Night DN is idle, only the first call is presented to it.

The Night DN may be defined as a multiple appearance DN with multiple call arrangement; all sets assigned the Night DN should be on the same node.

According to NAS routing, the Night DN defined on a node must be on the given node (local). If for any reason the Night DN is not on the local node Night Service Enhancements (NSE) are no longer supported.

In any case, NAS routing takes precedence over NSE, so if NAS routing is involved the call will be presented to the Night DN defined according to the NAS configuration.

If NAS routing is not involved and the Night DN defined on this node is located at a remote node (NSE no longer supported), the Night DN must be a remote Attendant DN to ensure calls are queued.

Night Service Network Environment

In network configurations with NAS routing, the Night Service Enhancements feature must be configured on each node in the network.

Feature interactions

Attendant Clearing during Night Service

The Night Service Enhancement features take precedence over Attendant Clearing during Night Service.

Attendant Interpositional Transfer

The requeuing of interpositional calls is not allowed. Night Service enhancements do not apply to interpositional calls, which remain on the console until answered.

Attendant Overflow Position

If a call with a ringing party on the destination side is presented at the last-active Attendant Console, and there is an active Attendant Overflow Position, then the ringing destination will be disconnected when the call is requeued. Likewise, if the call is a Call Waiting recall, Call Waiting will be canceled.

Call Forward All Calls
Position Busy
Attendant Forward No Answer

Any call which has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Call Waiting
Call Forward All Calls
Hunting
Call Forward Busy

Call Waiting, Call Forward All Calls, Hunting, and Call Forward Busy (for DID calls only) all take precedence over Camp-on.

Call Waiting will be applied by Night Service Enhancements to terminate incoming Night calls to busy Night DNs. This will still be done even if the Night DN is an analog (500/2500 type) telephone with Call Waiting Denied (CWD) Class of Service, or if the Night DN is a Meridian 1 proprietary telephone without a Call Waiting (CWT) key assigned.

All telephones will be given Night Call Waiting tone, if the NWT prompt in LD 15 was responded to with "YES," regardless of the Warning Tone (WTA/WTD) Class of Service setting of the telephone. Meridian 1 proprietary telephones will be given Night Call Waiting tone in the handset instead of the speaker buzz given for Call Waiting.

Call Waiting Redirection

Night Service has the same interaction with the Call Waiting Redirection feature as attendant-extended calls. Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also has precedence over the Call Waiting recall timer.

Centralized Attendant Service

Centralized Attendant Service (CAS) takes precedence over Night Service. If a user in a remote node in Night Service deactivates CAS and Camps-on an external call from the night station to a busy DN, and then reactivates CAS, any subsequent Camp-on recalls will be routed to the remote DN.

Dial Impulse Analog (500/2500 type) Telephone

A Dial Impulse analog (500/2500 type) telephone station must have TSA Class of Service to perform a station Camp-on.

Direct Inward System Access

It is not possible to assign a Night Service Group Number to any trunk that is a member of a route that is set to auto-terminate on a Direct Inward System Access DN.

Interposition Attendant Calls

This enhancement does not apply to interposition calls, which remain on the console until answered. The requeuing of interpositional calls is not allowed.

Network Attendant Service
Centralized Attendant Service
Attendant Overflow Position

Network Attendant Service (NAS) is mutually exclusive with Centralized Attendant Service and Attendant Overflow Position. The routing configuration for NAS will apply during Night Service. External calls and recalls may be queued to a remote Night DN, if defined. Internal calls and internal recalls queued during Day Service will be dropped, if the Night DN has been defined on a remote node.

For Camp-on from Inquiry Calls, NAS must be equipped at each node of the network.

Night Service

When the Night Service key is pressed on any Attendant Console, the customer enters Night Service and all Attendant Consoles are made Position Busy. It is then necessary to check all consoles for presented, but unanswered calls which must be cleared and requeued.

Recall with Priority during Night Service, Network Wide

If Recall with Priority during Night Service is equipped along with either the Night Service Improvement or Enhanced Night Service feature, calls are processed according to priority.

Trunk to Trunk Connection

Recalls made while the attendant is in Night Service are routed to the Night DN, if the original call is an external call. In such a case, the destination party is disconnected, the internal network trunk is released and the original extended call is presented to the Night DN. If the original call is internal, recalls are put in the attendant call waiting queue when in Night Service.

Feature packaging

The All Calls Remain Queued for Night Service, Recall to Night DN, and Requeuing of Attendant Presented Calls Night Service Enhancements are packaged as part of the International Supplementary Features (SUPP) package 131 for standalone applications. For network applications, the requirements are the International Supplementary Features (SUPP) package 131 and the Network Attendant Service (NAS) package 159 and its prerequisites.

For standalone Camp-on from Inquiry Call (Station Camp-on) applications the requirements are the Station Camp-on (SCMP) package 121 and the International Supplementary Features (SUPP) package 131.

For network Camp-on from Inquiry Call (Station Camp-on) applications the requirements are the Station Camp-on (SCMP) package 121, the International Supplementary Features (SUPP) package 131 and the Network Attendant Service (NAS) package 159 and its prerequisites.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – This overlay is modified to accept responses to STCB (Station Camp-on Busy Tone) and NSCP (Network Station Camp-on) prompts. In response to the STCB prompt, enter YES or NO to allow or deny Station Camp-on Busy Tone. In response to the NSCP prompt, enter YES or NO to allow or deny Network Station Camp-on, on a particular node.
2. LD 21 – This overlay is modified to print the STCB and NSCP prompts and their responses when the Customer Data Block is printed.
3. LD 22 – This overlay is modified to print the SCMP package mnemonic if the Station Camp-on package (121) is equipped.

LD 15 – This overlay is modified to accept responses to STCB (Station Camp-on Busy Tone) and NSCP (Network Station Camp-on) prompts. In response to the STCB prompt, enter YES or NO to allow or deny Station Camp-on Busy Tone. In response to the NSCP prompt, enter YES or NO to allow or deny Network Station Camp-on, on a particular node.

The prompt STCB will be output only if the SCMP (121) package is equipped. The prompt NSCP will be output only if the SCMP (121) and NAS (159) packages are equipped. By default, these two prompts will be set to NO.

Prompt	Response	Description
REQ:	CHG NEW	Change, or add.
TYPE:	FTR	Features and options.
...		

- STCB	(NO), YES	Station Camp-on Busy tone. Enter NO if Busy Tone is not to given to the transferring (controlling) party when the desired station is busy. Enter YES if Busy Tone is to be given to the transferring (controlling) party when the desired station is busy. The default is NO.
- NSCP	(NO) YES	Network Station Camp-on. Enter NO if sets on this node are not allowed to have calls camped-on by sets in other nodes. Enter YES if sets on this node are allowed to have calls camped-on by sets in other nodes. The default is NO.

LD 21 – This overlay is modified to print the STCB and NSCP prompts and their responses when the Customer Data Block is printed.

The STCB prompt and its response will be output only if the SCMP (121) package is equipped. The NSCP prompt and its response will be output only if the SCMP (121) and NAS (159) packages are equipped.

LD 22 – This overlay is modified to print the SCMP package mnemonic if the Station Camp-on package (121) is equipped.

Feature operation

Prior to describing the operation of the Night Service Enhancements the following terms are defined in terms of these feature operations:

Night DN

The Night DN pertains to Night DNs defined on a customer basis.

According to NAS routing the Night DN defined on a node must be on the given node (local).

External Call

Any call originated by the Public Switched Telephone Network (PSTN) is said to be an external call. This includes the following cases:

- Calls originating on a Public Exchange (Central Office [CO]), Foreign Public Exchange (FEX), Direct Inward Dial (DID), or Wide-area Telephone Service (WATS) trunk and terminating on the local node, and
- Calls originating on a CO, FEX, DID, or WATS trunk on a remote node, Integrated Services Digital Network (ISDN) TIE trunks, and NAS routed Public Switched Telephone Network (PSTN) or ISDN TIE trunks which are handled at the NAS node.

Non ISDN TIE trunks (local and remote) are said to be private trunks and are not treated as carrying external calls, although we may have a PSTN call involved at the originating node.

This definition includes both the standalone and network cases.

Requeuing of Attendant Presented Calls

Prior to the introduction of the Requeuing of Attendant Presented Calls, when a call had been presented to an Attendant Console it remained presented on the console, even if the Position Busy key was pressed.

The Requeuing of Attendant Presented Calls capability changes the system operation such that, if the Position Busy key is pressed on the console when an unanswered call has been presented to it the call will be returned to the attendant queue as if an AFNA time out had occurred.

This capability will not apply if the call is an interposition attendant (attendant to attendant) call. In this case, the call will remain on the console until answered.

In cases where the console is the last active console of the customer and there is an active AOP, if the call involves a ringing party on the destination side, the ringing will be disconnected. Similarly if the call is a Call Waiting recall, the Call Waiting will be canceled. This ensures that the required call will be presented on the AOP, irrespective of normal call type restrictions.

Note that all consoles will enter the Position Busy state if the Night Service key is pressed on any one of a customer's Attendant Consoles. In this case, all consoles must be checked for presented, but unanswered calls which must be cleared from the console and requeued.

Call Handling in Night Service

Calls already Queued when Night Service is Entered

Standalone case

Any external call which is queued, waiting to be serviced by an Attendant Console, when a customer goes into Night Service will continue to be queued until it can be presented to the appropriate Night DN.

Network case

As NAS takes precedence over NSE, if NAS routing is involved, the call will be presented to a remote attendant, or remote Night DN, or local Night DN, according to the NAS configuration.

If NAS routing is not involved, the call will be presented to local Night DN.

External Calls already Queued when Night Service is Entered

Operation Prior to Night Service Enhancements

The treatment of queued external calls was as follows:

- Dial "0" calls from DIDs or incoming CO calls remained queued for the Night DN.
- Call Forward Busy calls remained queued for the Night DN.
- Call Forward No Answer calls were not queued for a busy Night DN. If a call could not be presented immediately it was removed from the queue and the originating party was given Busy Tone.
- Attendant Recalls (ARC) and transfers to the attendant DN were removed from the attendant queue. The consultation call was "canceled", if the held call was an external party it was reconnected to the transferring (controlling) party.

- All intercepts involving an external party were queued for the Night DN.
- Timed reminder recalls remained queued, but were not presented to the Night DN.

Operation with Night Service Enhancements

The NSE capabilities change the operation such that Call Forward No Answer calls, ARCs, and transfers to the attendant will remain queued for the Night DN. In addition to these call types, timed reminder recalls will also be presented to the appropriate night DN.

Timed reminder recalls treatment is the following:

- Ringing stops for slow answer recalls when the recall occurs.
- Call Waiting is canceled when the recall occurs.
- Camp-on is canceled when the recall occurs.

Internal Calls already Queued when Night Service is Entered

Operation Prior to Night Service Enhancements

Any internal call that was already queued for the attendant was not queued for the Night DN.

When a customer went into Night Service, if the Night DN was idle, the first call was presented to the Night DN. Any internal calls not presented in this way were given busy tone and removed from the queue.

Operation with Night Service Enhancements

Standalone case

With NSE the operation is changed such that all internal calls which should be presented to the Night DN will remain queued until the customer Night DN becomes available.

Network case

If the call was extended by the attendant over DPNSS1 or MCDN with NAS active, and the call is camped-on or call waiting at the remote node, the call will remain queued at the local node waiting for an answer at the remote node.

Timed Reminder Time Outs during Night Service

When a timed reminder time out occurs during Night Service, depending on the call type, the call may be presented to the Night DN or continue waiting for the called party to answer. External (PSTN originated) calls will be presented to the Night DN or, if the Night DN is busy will wait in the queue until the Night DN becomes available.

In the case of a timed reminder Camp-on recall, the Camp-on is canceled when the recall occurs (time out).

In case of a slow answer recall, the desired set will be disconnected when the recall occurs (time out).

In case of a timed reminder Call Waiting recall, the Call Waiting will be canceled when the recall occurs (time out).

According to NAS routing these calls may be presented to a remote attendant or a remote Night DN. When the NAS routing starts, the destination (desired party) is released and the call is presented or queued to the appropriate terminal (that is, remote attendant or local Night DN or remote Night DN).

External calls that recall will be presented to, or queued for, the Night DN.

Internal calls that recall will be dropped when NAS routing is involved and the Night DN is at a remote node, because when NAS routing takes place internal call recalls are not queued for the Night DN. The station to which the call is being transferred (that is, the station on which the call is ringing, Call Waiting or camped-on) does not have to be located on the same node as the transferring (controlling) station.

If the attendant on the same node as the Night DN comes back to Day Service, timed recalls queued for the Night DN will be presented to the attendant as recalls.

Camp-on from Inquiry Call (Station Camp-on)

Standalone case

Any station, not necessarily the Night DN, attempting to transfer an external call, may, during the associated inquiry call, camp the trunk on to a busy station.

The camp-on will take affect from the moment the transferring station has completed the transfer to the desired DN.

The transferring station will hear Ringback Tone or Busy Tone depending on the option entered in response to the STCB prompt in the Customer Data Block (LD 15). This prompt applies to any set, not just the Night DN. By default (STCB is set to NO), the transferring party will hear Ringback Tone.

The desired station will hear Camp-on tone if it has WTA Class of Service assigned. Otherwise, if it has WTD Class of Service, the Camp-on will take effect without the desired party being informed a call is camped-on.

When the transfer is completed, the external party is camped-on to the desired station and receives either ringback tone or an announcement.

Network case

Any station, not necessarily the Night DN, attempting to transfer an external call across an ISDN network may, during the associated inquiry call, Camp-on the trunk on to a busy station.

The location of the transferring party has no effect on the Station Camp-on capability.

The Camp-on will take Affect from the moment the transferring station has completed the transfer to the desired DN.

The transferring station will hear ringback tone or busy tone depending on the option entered in response to the STCB prompt in the Customer Data Block (LD 15). This prompt applies to any set, not just the Night DN. By default (STCB is set to NO), the transferring party will hear ringback tone. The tone given, either ringback tone or busy tone, is determined by the node in which the desired (Camped-on to) party resides.

The desired station will hear Camp-on tone if it has WTA Class of Service assigned. If it has WTD Class of Service, the Camp-on will take affect without the desired party being informed a call is camped-on.

When the transfer is completed, the external party is camped-on to the desired station and receives either ringback tone or an announcement.

Recall Timing on Camp-on Calls

When any station extends an external call, recall timing will be initiated if the call is camped-on to a busy station.

The recall timing will start from the moment that the extending station "releases" the call. The value of the recall timer is set by the prompt RTIM in the Customer Data Block (LD 15).

At the recall, the Camped-on call will be routed to the attendant. If the attendant is in Night Service, night treatment is given, and if NAS routing is active, the call will be routed according to the NAS configuration.

Standalone case

When the recall to the attendant occurs, the Camp-on is canceled. If the attendant is busy during the recall, the recall will be queued.

Network case

When the recall occurs and the attendant has answered the recall, the call will still be camped-on to the desired party. If during the recall the attendant is busy, the recall will be queued.

Night Service, Enhanced

Contents

This section contains information on the following topics:

Feature description 131
Operating parameters 132
Feature interactions 133
Feature packaging 134
Feature implementation 135
Feature operation 136

Feature description

This feature modifies the existing Night Service operation by allowing Public Network (Central Office [CO], Direct Inward Dialing [DID], Foreign Exchange [FEX], and Wide Area Telephone Service [WATS]) trunks to be assigned to specific Directory Numbers (DN) during Night Service.

With this feature each customer will be able to assign Public Network trunks to one of nine Night Groups. Each Night Group will allow the customer to define up to nine Night DNs. During Night Service, incoming calls will be routed to one of the Night DNs defined for the group. The actual DN the call will be routed to is determined by the Night Service Option number selected at that time.

The customer will also be able to define whether Night Call Waiting tone will be given to Night stations. With Night Call Waiting tone allowed, busy Night stations are notified when an incoming call is terminating on them. The incoming call will be queued on the Night station until it becomes idle. When the Night station becomes idle, the incoming call will be presented.

This enhancement allows incoming DID trunks to be queued against busy Night stations, thereby making their operation the same as all other Public Network trunks.

Normal Night Service

With the feature active, the existing Night Service feature is enhanced by providing a night (NITE) prompt for DID trunks. Night numbers for DID trunks can be defined in their respective trunk blocks against the prompt. Attendants will be able to change their night numbers by specifying their corresponding access codes and member numbers using the existing Night Service feature.

Group Night Service

The customer is allowed to assign individual Public Network trunks to one of nine Night Group numbers (1 to 9). Each Night Group has up to nine Night Directory Numbers associated with it. During Night Service, incoming calls on a trunk will be routed to one of the Directory Numbers associated with that trunk. The actual number called is determined by a Night Service Option number corresponding to the Night Group number programmed by the attendant during Day service.

When an incoming call is routed to a busy directory number, an optional Night Call Waiting tone may be applied to that number to notify the user that a call is waiting. The call on the trunk will be queued until the night directory number becomes free.

Operating parameters

The same feature requirements apply as for Night Service.

Enhanced Night Service does not apply to auto-terminate trunks.

Enhanced Night Service is permanently activated if the system has no attendant and the ENS option is set to "YES." In this case, the Night Service Option number can only be programmed in the Customer Data Block (LD 15).

Enhanced Night Service uses one Speed Call list as the Night Number Table.

The operation of the optional Night Call Waiting Tone is the same as Call Waiting Tone.

Night Service Option 0 and Night Service Group 0 are reserved for the customer Night number and should not be programmed in LD 18.

Feature interactions

AC15 Recall: Timed Reminder Recall

The Night Service Enhancements feature is used to direct the call to the Night DN if the original call is an external call and the SUPP package 131 is equipped. When there is an AC15 recall and the attendant is in Night Service, the called party is disconnected (the AC15 trunk is released) and the original call is presented to the Night DN.

Call Waiting (CWT)

This feature will terminate incoming Night calls to busy Night DNs by applying Call Waiting. This will still be done even if the Night DN is an analog (500/2500 type) telephone with Call Waiting Denied (CWD) Class of Service, or if the Night DN is a Meridian 1 proprietary telephone without a Call Waiting (CWT) key assigned.

All telephones — both analog (500/2500 type) telephones and Meridian 1 proprietary telephones — will be given Night Call Waiting tone, if the NWT prompt in LD 15 was responded to with "YES," regardless of the Warning Tone (WTA/WTD) Class of Service setting of the telephone. Meridian 1 proprietary telephones will be given Night Call Waiting tone in the handset, instead of the speaker buzz given for Call Waiting.

Direct Inward System Access (DISA)

It is not possible to assign a Night Service Group Number to any trunk that is a member of a route that is set to auto-terminate on a DISA DN.

Multi-Party Operations

During Night Service, mishandled calls are routed to the night DN. External calls, other than DID calls, are queued until answered. TIE calls are disconnected if the night DN is busy.

Multi-Tenant service

Any restrictions that exist in the system preventing individual Tenant access to certain routes will not be checked when the Night Number Table is programmed. It will be up to the technician to ensure all such restrictions are taken into consideration.

The tenant to route restrictions will be enforced when an attempt is made to terminate an incoming call on a Night DN via the Night Number Table. If the termination to the Night DN is not allowed, Overflow tone (Fast Busy) will be given to the incoming trunk.

Trunk Barring (Telephones)

Any incoming trunk call that is routed by Enhanced Night Service to a telephone from which it is barred will not be connected. Overflow tone (Fast Busy) will be given to the incoming trunk instead.

Trunk to Trunk Barring

Any incoming trunk call that is routed to an outgoing Public Network trunk will be barred if Enhanced Night Service is active. Overflow tone (Fast Busy) will be given to the incoming trunk instead. This restriction is in addition to the configured Trunk Barring for the system.

Warning Tone

All telephones — both analog (500/2500 type) telephones and Meridian 1 proprietary telephones — will be given Night Call Waiting tone, if the NWT prompt in LD 15 was responded to with “YES,” regardless of the Warning Tone (WTA/WTD) Class of Service setting of the telephone.

Feature packaging

Enhanced Night Service (ENS) is packaged as package 133.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 18 – Configure Night Number Table.
2. LD 15 – Configure Enhanced Night Service.
3. LD 14 – Configure Enhanced Night Service for trunks.

LD 18 – Configure Night Number Table.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	SCL	Speed Call List number
LSNO	xxx	List number. Enter list number; this number will be entered in response to the NNT prompt in LD 15 (Customer Data Block).
DNSZ	xx	Enter maximum excepted length required.
SIZE	100	Enter 100 to ensure that definitions for Options 1-9 and Groups 1-9 may be input.
STOR	xy z...z	Define Night Number Table entry, where: x is the Night Service Option number (1-9) y is the Night Service Group number (1-9), and z...z is the DN to which calls will be routed. This must be a valid station DN within the system. Network Access Codes are not allowed. ***Note:*** Night Service Option 0 and Night Service Group 0 are reserved for the customer Night number and should not be programmed, (that is, 00, 01, 02, 03, 04, 05, 06, 07, 08, 09, 10, 20, 30, 40, 50, 60, 70, 80, and 90).

LD 15 – Configure Enhanced Night Service.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	CDB	Customer data block.
...		
ENS	(NO) YES	(Disable) enable Enhanced Night Service.
- NWT	(NO) YES	(Disable) enable Night Call Waiting tone.
- NNT	0-253	Enter the Speed Call List (LSNO) number of the Night Number Table defined in LD 18.
- NSO	0-9	Night Service Option number.

LD 14 – Configure Enhanced Night Service for trunks.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dial.
...		
NGRP	(0)-9	Night Service Group number.

Feature operation

Night number assignment from Night Number Table

A Speed Call List (SCL) is specified in the Customer Data Block (CDB), LD 15, for the purpose of storing night DNs against each Night Service Group and Option.

The designated SCL consists of 100 two-digit translations. The first digit represents the Night Service Option number, while the second digit represents the Night Service Group number. Night Service Option zero (0) and Group zero (0) are reserved for the customer Night number, and therefore should not be defined, (that is, 00, 01, 02, 03, 04, 05, 06, 07, 08, 09, 10, 20, 30, 40, 50, 60, 70, 80, and 90). The following is a sample Night Number Table with an explanation of how calls are terminated:

Table 2
Example of a Night Number Table

Option	Group	Number
.	.	
.	.	
.	.	
2	5	4311
2	6	4011
2	7	3893
.	.	
.	.	
3	5	3400
3	6	4321
3	7	4780
.	.	
.	.	

Night stations 4311, 4011, 3893 are assigned to Night Service Option 2 for Night Service Groups 5, 6, and 7 respectively.

If Night Service Option 2 is active, night calls from trunks designated in LD 14 as Night Service Group 5 will be routed to 4311, night calls from trunks designated in LD 14 as Night Service Group 6 will be routed to 4011, and night calls from trunks designated in LD 14 as Night Service Group 7 will be routed to 3893.

If the attendant selects Night Service Option 3, night calls from trunks designated in LD 14 as Night Service Group 5 will be routed to 3400, night calls from trunks designated in LD 14 as Night Service Group 6 will be routed to 4321, and night calls from trunks designated in LD 14 as Night Service Group 7 will be routed to 4780.

Attendant Console

This section describes the sequences to be followed by the attendant to select and query the Night Service Option and to activate Enhanced Night Service.

Step	Action	Response
1	Press Shift key	
2	Press Loop key	Indicator is activated.
3	Press Night key	Indicator flashes. Dial tone is received. Current Night Service Option number is displayed.
4a	QUERY ONLY	
	Press RLS key	Indicator next to Loop and Night keys deactivates. Display is cleared.
or		
4b	SELECT	

i	Dial a one-digit (0-9) option number.	Dial tone is removed. Old Night Service Option number (X) is shifted, new Option number (Y) is displayed, and X and Y are separated by a hyphen, (for example, Y-X).
ii	Press RLS key	Indicator next to Night and Position Busy keys deactivates. Night Service Option is stored. Display is cleared.
5	ACTIVATE Enhanced Night Service Press Night key or Position Busy key if you are last active Attendant.	Indicators next to Night and Position Busy keys are activated. Current (active) Night Service Option number is displayed.

No Hold Conference

Contents

This section contains information on the following topics:

Feature description 141
Operating parameters 143
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Feature packaging 146
Feature implementation 146
Feature operation 148

Feature description

Combined with Conference, Speed Call, System Speed Call, Autodial, and Hot Line, No Hold Conference (NHC) allows you to establish a Conference call without placing the current caller on hold.

This feature is available in four forms, merging No Hold Conference (NHC) with Autodial, Speed Call, and Hot Line into a single key. The new combined keys are the Conference-Autodial (CA), Conference-Speed Call (CS), and Conference-Hot Line (CH) feature keys. A No Hold Conference (NHC) key can also be configured, acting as a simple Conference key.

Conference-Hot Line can be used in the following two ways:

- The Direct CH option has the number stored with the key.
- The List CH option has a pointer that selects an entry from a Hot Line list.

When a telephone is connected to another party, you can originate a Conference-Autodial (CA), Conference-Speed Call (CS), or Conference-Hot Line (CH) call by pressing the CA, CS, CH, or NHC key. The system determines the destination as if it were a regular Autodial, Speed Call, or Hot Line call. The parties are conferenced in without holding. For example, a call comes in to the customer notifying the customer of a fire. The user wishes to notify the fire department of the emergency without placing the original caller on hold, and the number is stored on the Conference-Autodial key. By pressing the CA key, the customer establishes a Conference call. The fire department is notified and the original connection is maintained.

When you press the feature key, one of the following occurs:

- If the destination is an idle internal Directory Number (DN), that DN rings and the CA, CS, CH, or NHC lamp flashes (60 ipm). You hear no ringback tone.
- If the destination is a trunk with answer supervision, the trunk is seized and the key lamp flashes. The voice path is not established until an answer signal is received.
- When the destination is a trunk without answer supervision, the trunk is seized, the voice path is established, and the key lamp flashes. All tone signals provided by the far end (for example, ringback) are heard by all parties involved in the Conference call. Calls on trunks without answer supervision are treated as answered after digit outpulsing is completed.
- When the intended destination is a busy internal DN, trunk, or route, the key lamp fast flashes (120 ipm). Press the active call key to cancel the attempt. The active call key is the key on which the call is established. It can be any key on which a regular Conference call can be made, including the DN key, Call Waiting, and Automatic Call Distribution (ACD) Incalls keys.
- In the case of network blocking, or if a conference port is unavailable, the key lamp fast flashes. Press the active call key to cancel the attempt.
- When the destination is an invalid entry (for example, a vacant number, or an illegal list entry) the key lamp fast flashes. Press the active call key to cancel the attempt.

Pressing the active call key at any time before the called party responds cancels the attempt, returning the telephone to the state prior to pressing the CA, CS, CH, or NHC key.

If the call is answered, the key lamp goes off, and the called party is added to the existing conversation. By pressing the active call key, the last added party is released. These operations can be repeated as often as necessary, according to your network configuration, to add new parties to an existing conversation.

If the CA or CS keys are pressed at any time other than during a Conference call, they operate as a regular Autodial or Speed Call. The CH key operates as a regular Hot Line key only when the terminating key is HOT. Pressing the NHC key allows the user to dial the number desired for the Conference call.

Operating parameters

Assignable keys are limited to the number of keys available on your telephone.

NHC is available on Meridian 1 proprietary telephones with the CA, CS, CH, and NHC keys. It is not available on the M3000, analog (500/2500 type) telephones, or Attendant Consoles.

The Release (RLS) key has no effect while the key lamps are flashing or fast flashing. Other than during these stages, it can be used to abort the Conference call.

The CA key, like the regular Autodial key, is programmable from the telephone.

The CS and CH keys must have the Speed Call and Hot Line numbers assigned in LD 18.

Data calls are not supported.

All four keys can coexist with each other as well as with other Conference, Autodial, Speed Call, and Hot Line features.

Feature interactions

500/2500 Line Disconnect

If one of the parties in the conference is connected to a 500/2500 port that is in turn connected to a Voice Response Unit (VRU), dial tone is provided to the 500/2500 port when all the other parties in the conference disconnect. This feature enhancement applies in the same way to Call Transfer and Hunting.

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the No Hold Conference (NHC) key is ignored.

Automatic Hold

The Conference-Hot Line (CH) key does not support Automatic Hold.

Call Page Network Wide

A station set or Attendant Console that no hold conferences an external Call Page Network Wide (PAGENET) uncontrolled call is not blocked. However, an external PAGENET controlled call is blocked.

Centralized Attendant Services

Centralized Attendant Service (CAS) attendants are not supported.

Conference - Six Party

This feature can be enabled at any time that a regular Conference-6 feature can be activated.

Display of Calling Party Denied

Display information on sets involved in a No Hold Conference call is based on the individual Class of Service of each set.

Hot Line

The CH key supports only one-way Hot Line calls.

IVR

IVR calls cannot be No Hold conferenced.

Make Set Busy

The CH key overrides Make Set Busy only when the terminating key is HOT.

Meridian 911

In a Meridian 911 environmental, No Hold Conference calls are treated as internal calls and are linked to the low priority queue of the ACD DN.

Meridian 911 Call Abandon

M911 abandoned calls cannot be No Hold conferenced.

Music Trunk

A Music (MUS) Trunk cannot be No Hold conferenced.

Recorded Announcement Trunk

A Recorded Announcement (RAN) Trunk cannot be No Hold conferenced.

Off-Hook Alarm Security

Off-Hook Alarm Security treatment occurs when a telephone with ASCA Class of Service attempts an NHC call and the ASTM expires. The OHAS DN is conferenced in with the other conferees.

Paging Trunk

A Paging (PAG) Trunk cannot be No Hold conferenced.

System Speed Call list

Whenever the CS key is programmed for a System Speed Call list, all calls made with that key are System Speed Calls.

Feature packaging

No Hold Conference capability is available when the following features are equipped:

- Autodial (ADL) for CA key configuration
- Speed Call User (SCU) if the CS key is configured
- Enhanced Hot Line (HOT) package 70 for the CH key, and
- System Speed Call (SSC) package 34 to configure CS or CH keys.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 18 – Provision Speed Call or Hot Line numbers for CS and CH keys.

2 LD 11 – Add or change No Hold Conference for Meridian 1 proprietary telephones.

LD 18 – Provision Speed Call or Hot Line numbers for CS and CH keys.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change a Speed Call list.
TYPE:	SCL SSC HTL	Speed Call, System Speed Call, Hot Line.
CUST	xx	Customer number, as defined in LD 15
LNSO	0-8190	Speed Call list number.
NCOS	(0)-99	NCOS (when TYPE = SSC or HTL).
DNSZ	xx	Maximum number of digits in a list entry, where: xx = 4, 8, 12, (16), 20, 24, 28, or 31.
SIZE	1-1000	Maximum number of entries in the Speed Call list.

WRT	(YES) NO	Data is correct and list can be updated.
STOR	xxx	xxx = list entry number (0-9, 00-99, or 000-999).
	yy...yy	yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	NO (YES)	Data is correct and list can be updated.

Note: The WRT prompt follows the SIZE and STOR prompts asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating that the data was not stored and must be reentered.

A response of (*) aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved.

The following information is displayed with the WRT prompt, following SIZE: ADDS: MEM: xxxxx DISK: yy.y

Where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new speed call list. Check the MEM AVAIL and DISK REC AVAIL values displayed before the REQ prompt.

LD 11 – Add or change No Hold Conference for Meridian 1 proprietary telephones.

Prompt	**Response**	**Description**
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

KEY	xx CA 4-(16)-23 y...y	Combined NHC and Autodial key, where: xx = key number, and y...y = target number stored in the key (maximum 23 digits).
	xx CH D yy z...z	Combined NHC and Direct Hot Line key, where: xx = key number yy = number of digits in the target number, and z...z = target number stored within the key.
	xx CH L 0-999	Combined NHC and Hot Line key, where: xx = key number, and 0-999 = Hot Line list entry.
	xx CS yyy	Combined NHC and Speed Call key, where: xx = key number, and yyy = Speed Call list number.
	xx NHC	NHC key, where: xx = key number.

Feature operation

No Hold Conference (NHC)

To establish an NHC call using the NHC key:

1. Establish a call.
2. Press **NHC**. The indicator goes on steadily.
3. Dial the number for the conference. The indicator flashes until the call is answered.
4. The conference is complete.

Conference-Autodial (CA)

To store an Autodial number:

1. Press **CA** (Conference-Autodial). The CA indicator flashes.
2. Enter the number.
3. Press **CA**. The indicator goes off.

To use Conference-Autodial:

1. Establish a call.
2. Press **CA**. The indicator flashes until the call is answered.
3. The conference is complete.

Conference-Hot Line (CH)

To establish an NHC call using the CH key:

1. Establish a call.
2. Press **CH** (Conference-Hot Line). The indicator flashes until the call is answered.
3. The conference is complete.

Conference-Speed Call (CS)

To establish an NHC call using the CS key:

1. Establish a call.
2. Press **CS** (Conference-Speed Call). The indicator goes on steadily.
3. Enter the Speed Call list entry number for the conference number. The indicator flashes until the call is answered.
4. The conference is complete.

Note: To disconnect the last NHC conference caller in any of the above procedures, press the DN key once.

North American Numbering Plan

Contents

This section contains information on the following topics:

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Feature description

The North American Numbering Plan (NANP), established in 1947 and currently administered by Bellcore, governs the telephone numbering system throughout Bermuda, Canada, the Caribbean, and the United States.

Two components of the NANP are Interchangeable Numbering Plan Areas (INPAs) and Carrier Access Codes (CACs). NPAs are the three-digit prefixes commonly known as area codes. CACs permit telephone users to access any interexchange carrier or operator service provider. CACs must be supported by any entity, such as a hotel, motel, hospital, university, airport, gas station, or pay telephone owner, that makes telephone services available to the public.

Interchangeable Numbering Plan Area

The Interchangeable NPA codes plan was developed in the 1960s to manage the inevitable depletion of available codes. Prior to 1995, all area codes had an N(0/1)X format, where N was any digit from 2 to 9 inclusive and X was any digit, 0 to 9. As of January 1995, area codes have an NXX format, increasing the available codes to 640.

Modifications to system software, including changes to LDs that accept NPA or Home NPA codes, have eliminated dependencies and limitations associated with the old NPA code format.

The introduction of Interchangeable NPAs means that an area code (NPA) can appear identical to a Central Office prefix or a private network Location Code (LOC).

It is important to avoid conflicts among NPAs, Central Office prefixes, and LOCs. It is recommended that customers implement 1+ dialing to eliminate ambiguity.

The remainder of this section discusses the procedure that Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) customers need to follow to handle the NPA changes. Although Alternate Route Selection (ARS) and Direct Trunk Access customers need not modify their databases, those who use Call Detail Recording and/or Toll Denied Class of Service should consider the effect of NPA changes on their operations.

BARS/NARS

BARS/NARS prohibits the entry of identical NPAs, Central Office prefixes, or LOCs. Typically, customers construct translation tables with NPA and LOC codes associated with one Access Code and Central Office codes associated with a second Access Code. Now that LOC and NPA codes may be identical, this option no longer guarantees that codes will not conflict.

Table 3 summarizes the options.

Table 3
Access Codes and 1+ dialing

# of Access Codes	Need LOC?	Use 1+?	Results
2	yes	yes	no conflicts
2	yes	no	may need to check that no LOC is identical to any NPA (depends on access code arrangement)
2	no	yes	no conflict
1	no	yes	no conflict
1	no	no	not recommended
1	yes	yes	not recommended

The ideal dialing plan continues to use two Access Codes, with 1+ dialing for NPA calls. (Digit Manipulation can remove the "1" for customers whose Central Office does not support 1+ dialing.)

Customers with two Access Codes that do not want to use 1+ dialing must ensure that no LOCS in the database are identical to existing NPAs. The database needs to be checked whenever a new NPA is introduced.

Customers who do not need LOCs can use a single Access Code and 1+ dialing or two Access Codes, one for NPA and one for the Central Office code.

Direct Trunk Access and Alternate Route Selection

Direct Trunk Access and Alternate Route Selection customers need not update software to support interchangeable NPAs. Customers using Direct Trunk Access should continue to monitor local dialing procedures to ensure correct toll call recognition.

System upgrades

Upgrade requirements can include hardware and software. For specific information, consult *Large System: Upgrade Procedures* (553-3021-258).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Autodial
Speed Call
Hot Line

Customers may need to modify the lists and tables associated with these features to accommodate the new prefixes or to reflect changes to numbers resulting from implementation of 1+ dialing.

Feature packaging

Equal Access compliance is included in base system software. The Network Class of Service package (NCOS) package 32 is required to configure Equal Access.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Change the Home Numbering Plan Area Code at the HNPA prompt.

2 LD 16 – Enter the NPA code definition for the M911 feature.

3 LD 19 – Enter the NPA for incoming Feature Group D ANI screening.

4 LD 87 – Define the Free Call Area Screening.

5 LD 90 – Build the NPA and HNPA translation tables.

Note 1: For complete information on implementation and configuration, refer to the Equal Access Compliance feature description in this document.

Note 2: The following prompts have been modified to accept NPA input in the new interchangeable format.

LD 15 – Change the Home Numbering Plan Area Code at the HNPA prompt.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	NET	ISDN and ESN Networking options.
...		
- ISDN	YES	Change ISDN options.
- HNPA	200-999 1200-1999	Home Numbering Plan Area code.

LD 16 – Enter the NPA code definition for the M911 feature.

Prompt	Response	Description
...		
TYPE	NPID	Numbering Plan Digit/Information Digit table.
IDTB	0-7	NPID table number.
NPID	0-9	NPID to be translated.
TRMT	NPA	NPID treatment.
NPA	200-999	Numbering Plan Area code.

LD 19 – Enter the NPA for incoming Feature Group D ANI screening.

Prompt	Response	Description
...		
TYPE	ANI	Feature Group D data block.
ANIT	(OVF) RAN xxx DN xxx NCOS xxx	Invalid Automatic Number Identification (ANI) treatment.
NPA	200-999	Three ANI digits in NPA format (prompt accepts only three digits even if 1+ dialing is in effect).

LD 87 – Define the Free Call Area Screening.

Prompt	Response	Description
...		
FCI	xxx	Free Call Area Screening table index number.
NPA	200-999 200-999 200-999	Area code or extended NPA code translation (only three digits accepted even if 1+ dialing is in effect).

LD 90 – Build the NPA and HNPA translation tables.

Prompt	Response	Description
...		
TRAN	AC1, AC2, SUM	Access code 1, 2, or summary tables.
NPA	200-999 200-999 200-999 1200-1999 1200-1999 1200-1999	Area code or extended NPA code translation.
HNPA	200-999 1200-1999	Home Numbering Plan Area code.

Carrier Access Codes

A Carrier Access Code (CAC) gives a caller access to any interexchange carrier or Operator Service Provider (OSP). FCC regulations require that Call Aggregators, such as hotels, motels, hospitals, universities, airports, gas stations, and pay telephone owners, provide selective access to the public. Callers dial the CAC to reach their desired carrier or OSP before dialing the telephone number.

Aggregators are permitted to block calls selectively, although they must allow callers access to any long distance caller. Selective equal access lets aggregators choose to block direct-dialed calls that result in charges to the originating telephone. Aggregators cannot block operator-assisted calls.

The CAC has included a “10” identifying prefix followed by a three-digit Carrier Identification Code (CIC) for a total of five digits. FCC regulations, require that the CAC expand to seven digits: a “101” identifying prefix followed by a four-digit CIC.

Feature operation

System software allows the following operator-assisted North American and international dialing sequences:

- CAC + 0
- CAC + 0 + (NPA) + NXX + XXXX
- CAC + 01 + CC + NN

System software allows or denies these direct-dialed calls:

- CAC + 1 + (NPA) + NXX + XXXX
- CAC + 011 + CC + NN

where

CAC = Carrier Access Code (10XXX or 101XXXX)

NPA = Numbering Plan Area (area code)

NXX = Central Office code format
(N = any digit except 0 or 1; X = any digit 0–9)

XXXX = any four digits

CC = Country Code, and

NN = National number.

Off-Hook Alarm Security

Contents

This section contains information on the following topics:

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Feature description

Off-Hook Alarm Security (OHAS) allows locked out calls to be intercepted to a customer-defined Directory Number (DN) other than an attendant (for example, a security DN). OHAS treatment is determined on a telephone basis by assigning a Class of Service called Alarm Security Allowed (ASCA). By enhancing line lockout, telephones with Alarm Security Allowed (ASCA) Class of Service are intercepted to customer-defined Directory Numbers (DNs) when the dial tone/interdigit timer expires or the telephone is Forced Out of Service (FSVC). Telephones without ASCA continue to use the existing line lockout treatment; refer to the Line Lockout module in this document.

An Off-Hook Alarm Security (OHAS) DN can be a Single Appearance Directory Number (DN), a Multiple Appearance DN, or an Automatic Call Distribution (ACD) DN. The OHAS DN cannot be an attendant DN, Listed DN, SPRE, Virtual ACD Agent, or Trunk Access Code.

If the ASCA Class of Service is assigned, but the telephone is not associated to an OHAS DN, an error message appears on the maintenance TTY when the system tries to redirect the call.

The Alarm Security Timer (ASTM) provides dial tone and interdigit timing for telephones with ASCA Class of Service. The ASTM does not apply to telephones being Forced Out of Service (FSVC).

Dial tone and interdigit timeout – call treatment

A telephone associated with an OHAS DN that receives a dial tone or interdigit timeout intercepts to the OHAS DN specified by the telephone's Off-Hook Interdigit OHAS number (OHID).

Forced Out of Service (FSVC) – call treatment

A digital telephone is considered FSVC when the line is cut, damaged, or unplugged.

The FSVC OHAS treatment applies only to digital telephones. A telephone associated with an OHAS DN that is FSVC intercepts to the OHAS DN specified by the telephone's FSVC number.

Multiple OHAS DNs

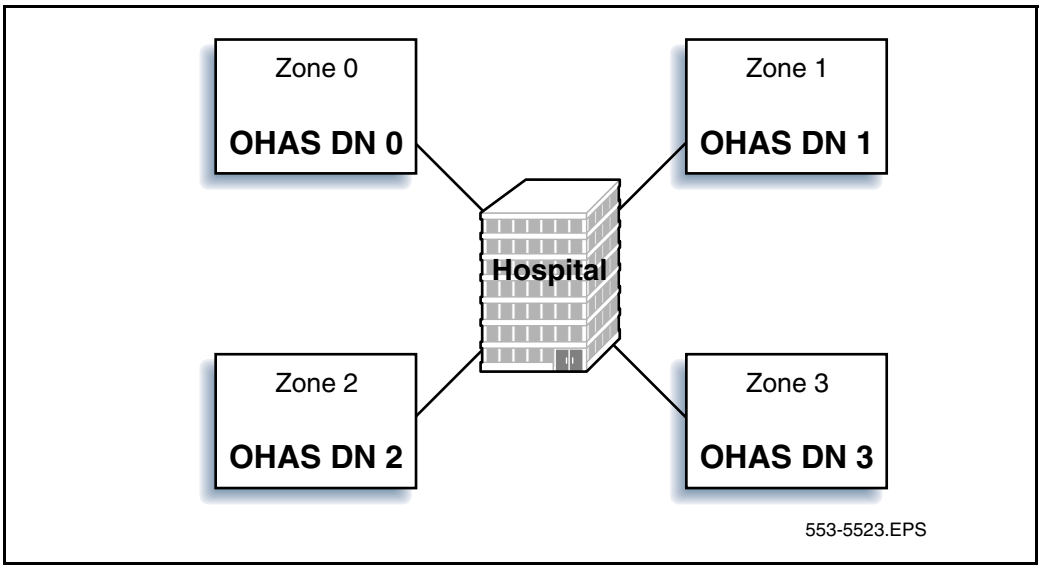
The two methods for handling multiple OHAS DNs are zone and event dependent, and are described in the following sections.

Multiple OHAS DNs – zone dependent

OHAS allows for multiple OHAS DNs within a single customer group, enabling the customer to create multiple zones.

For example, a hospital with several locations can define separate OHAS DNs for each location and define each distinct location as a zone. In Figure 2, the hospital has four zones. A separate OHAS DN is defined for each of the four zones. Zone 0 uses OHAS DN 0, Zone 1 uses OHAS DN 1, and so on. Each telephone in Zone 0 defines the OHID and FSVC numbers to 0; each telephone in Zone 1 defines the OHID and FSVC numbers to 1, and so on.

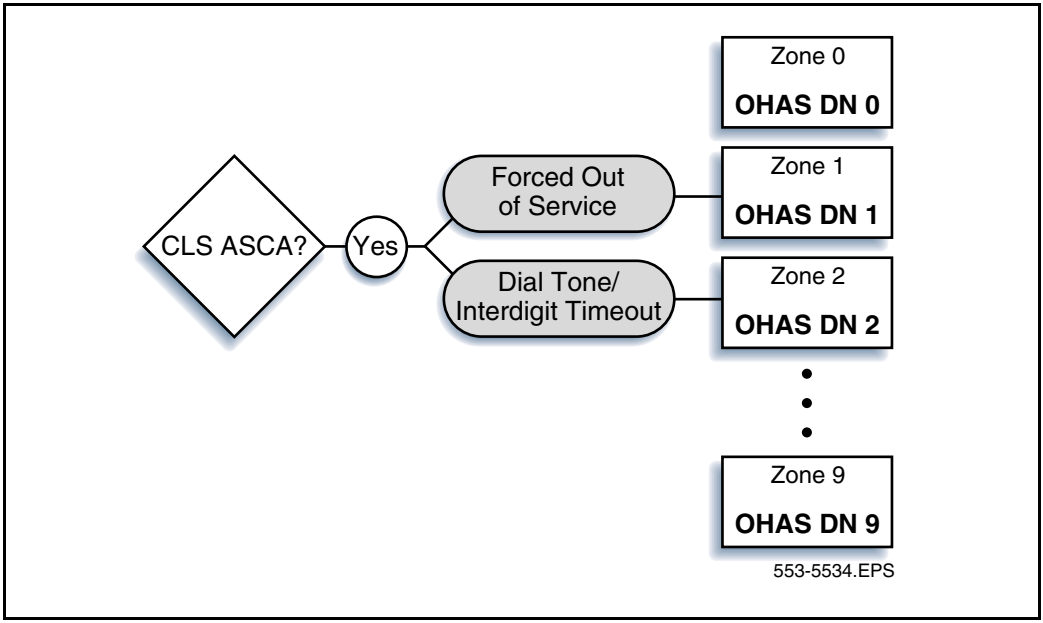
Figure 2
Zone dependent example



Multiple OHAS DNs – event dependent

OHAS can distinguish between OHID timeout and FSVC events by having a single telephone with separate OHAS DNs for OHID timeout and FSVC events (for example, a telephone can be defined with a FSVC number 1 and OHID number 2. If a dial tone/interdigit timeout occurs, the telephone intercepts to OHAS DN 2. If the same telephone is FSVC, OHAS DN 1 is notified).

Figure 3
Event dependent example



OHAS TTY display

Every time an OHAS intercept treatment takes place, a message is sent to all maintenance TTYs. This message contains an OHAS message indicator, the originating DN and TN, and a time stamp.

Format			
OHASxxxx	<dn>	l s c u	time stamp
Output example			
OHAS0000	5003	1 0 1 0	04:30:21

Note: The two possible OHAS messages are:
OHAS0000OHAS treatment due to dial tone/interdigit timeout, and
OHAS0001OHAS treatment due to Forced Out of Service call treatment.

Operating parameters

OHAS is not supported for attendants or networks.

OHAS intercept treatment for FSVC telephones is provided only for the following telephones:

- the M2009, M2112, and M2018
- the M2317
- the M3000, and
- the M2006, M2216, M2616, M2008, and M2016.

The Alarm Security Timer (ASTM) does not apply to telephones being FSVC.

The timing for recognizing a FSVC condition depends on the type of card that the system is using:

- The Integrated Services Digital Line Cards (ISDLCs) take approximately six seconds to recognize an FSVC condition.
- Peripheral Controller cards take approximately one second to recognize an FSCV condition.

Once a trunk is seized, OHAS treatment does not apply.

Feature interactions

Call Redirection

Call Redirection features defined for telephones with ASCA Class of Service work as currently defined in the system. The Call Redirection features include the following:

- Call Forward All Calls
- Call Forward No Answer
- Call Forward Busy
- Call Forward by Call Type

- Call Pickup, and
- Hunting.

Call Transfer

A telephone receives the OHAS treatment if the telephone has ASCA Class of Service and attempts to transfer a call and the ASTM expires.

China – Flexible Feature Codes - Busy Number Redial Enhanced Flexible Feature Codes

Busy Number Redial cannot be used on a set with Off-Hook Alarm Security Allowed, since ADL cannot be configured on these sets.

Conference

The OHAS line lockout treatment occurs when a telephone associated with an OHAS DN initiates a Conference call and the ASTM expires. Only the Conference initiator receives the OHAS treatment; other conferees remain in Conference. If the initiator of the Conference call presses the Conference key, the OHAS DN is conferenced in with the other conferees.

Electronic Switched Network Trunk Access Codes

If an Electronic Switched Network or Trunk Access Code is dialed, the dial tone/interdigit timer is stopped and the telephone will not recall to the designated OHAS DN after the specified time period has elapsed.

Last Number Redial Stored Number Redial

OHAS treatment may apply to these features if the ASTM expires.

Line Lockout

OHAS treatment occurs when a telephone with ASCA Class of Service receives an interdigit or dial tone timeout. The ASTM is used instead of the dial tone and interdigit timers (DIDT and DIND, respectively) normally used for LLT and DLT line lockout treatment.

Multi-Party Operations

Three-party Service (TSA) and Alarm Security Allowed (ASCA) Classes of Service are mutually exclusive. A set assigned TSA Class of Service cannot also be assigned ASCA Class of Service, and vice versa; a set assigned ASCA Class of Service cannot also be assigned TSA Class of Service.

The Off-Hook Alarm Security feature is mutually exclusive with Multi-Party Operations.

No Hold Conference

OHAS treatment occurs when a telephone with ASCA Class of Service attempts an No Hold Conference call and the ASTM expires. The OHAS DN is conferenced in with the other conferees.

Room Status

OHAS takes precedence over the off-hook detection method of the Room Status feature. If a telephone is defined with the Alarm Security Allowed (ASCA) Class of Service, the off-hook detection method does not work.

Speed Call
Speed Call, System

OHAS treatment may apply to these features if the ASTM expires. The Alarm Security Timer may expire for the following reasons:

- A dial tone or interdigit timeout occurs while dialing the speed call access code.
- The Speed Call being accessed has an asterisk (*) causing a three-second delay. If the ASTM is three seconds or less, the OHAS intercept treatment may occur.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Define the Off-Hook Alarm Security (OHAS) Directory Numbers (DNs). OHAS DNs must have ASCA Class of Service assigned in LD 10 or LD 11.
2. LD 10 – Assign Alarm Security Allowed (ASCA) Class of Service.
3. LD 11 – Assign Alarm Security Allowed (ASCA) Class of Service.

LD 15 – Define the Off-Hook Alarm Security (OHAS) Directory Numbers (DNs). OHAS DNs must have ASCA Class of Service assigned in LD 10 or LD 11.

Prompt	Response	Description
REQ:	NEW CHG	Add or change a customer.
TYPE:	INT	Intercept Treatment options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- LLT	(OVF) ATN OFA	Flexible line lockout treatment.
...		
TYPE	OAS	Off Hook Alarm Security Options.
The following prompts occur when OAS_DATA is entered:		
- ODN0	xxx...x	OHAS DN 0.
- ODN1	xxx...x	OHAS DN 1.
- ODN2	xxx...x	OHAS DN 2.
- ODN3	xxx...x	OHAS DN 3.

- ODN4	xxx...x	OHAS DN 4.
- ODN5	xxx...x	OHAS DN 5.
- ODN6	xxx...x	OHAS DN 6.
- ODN7	xxx...x	OHAS DN 7.
- ODN8	xxx...x	OHAS DN 8.
- ODN9	xxx...x	OHAS DN 9.
- ASTM	1-(30)-63	The timer applies to all OHAS DNs and is programmable in one-second increments.

LD 10 – Assign Alarm Security Allowed (ASCA) Class of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
CLS	(ASCD) ASCA	Alarm Security (denied) allowed. When ASCA is assigned, the OHAS DN must be defined in LD 15.
OHID	(0)-9	Off-Hook Interdigit OHAS number.

LD 11 – Assign Alarm Security Allowed (ASCA) Class of Service.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	aaaa	Telephone type, where: aaaa = 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
CLS	(ASCD) ASCA	Alarm Security (denied) allowed. When ASCA is assigned, the OHAS DN must be defined in LD 15.

OHID	(0)-9	Off-Hook Interdigit OHAS number.
FSVC	(0)-9	FSVC OHAS DN number. The FSVC prompt is given only to digital telephones.

Feature operation

No specific operating procedures are required to use this feature.

Off-Hook Alarm Security Half Disconnect Enhancement

Contents

This section contains information on the following topics:

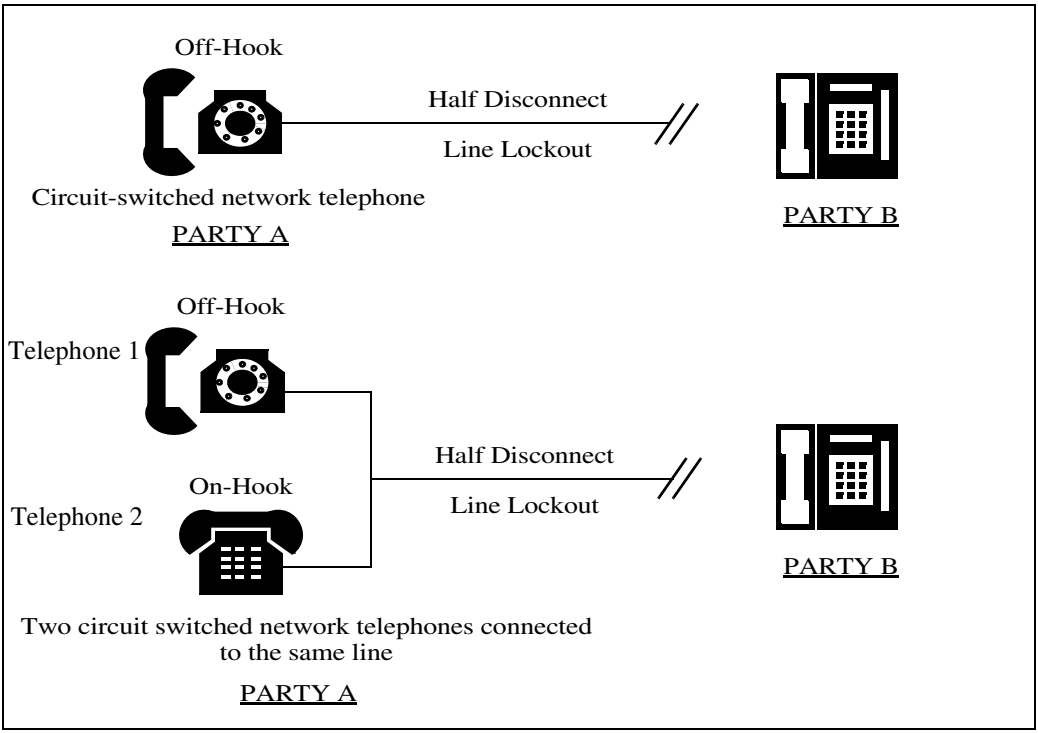
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Feature description

The Off-Hook Alarm Security Half Disconnect Enhancement (OHAS HD) feature, enhances the functionality of the existing Off-Hook Alarm Security (OHAS) feature. The existing Off-Hook Alarm Security (OHAS) feature allows a user to indicate an emergency by going off hook. The security DN programmed for the off-hook telephone rings after the dial tone/interdigit timer expires.

Where two telephones share one TN, the need for an enhancement addressing the Half Disconnect condition arose. The scenario is as follows. A user initiates a call on telephone 1 and then continues the call on telephone 2, in a different location. When the user completes the call, but only hangs up telephone 2, (telephone 1 remains off hook) the line (Party A) remains in the Half Disconnect/Line Lockout state until the user remembers to put telephone 1 on hook. (See Figure 4.)

Figure 4
OHAS HD Scenario



When the OHAS HD feature is enabled you can define, on a customer basis, the length of time before the OHAS HD treatment is given. When this timer expires the programmed security DN rings. If a telephone goes on-hook before the OHAS Half Disconnect Timer (HDTM) expires, the OHAS Half Disconnect treatment is canceled, as the telephone has completed its disconnect.

The OHAS Half Disconnect Option, (HDOPT) determines the number of OHAS Half Disconnect treatments that can be given to telephones that remain in the Half Disconnect state. This is programmed on a customer group basis.

There are three OHAS HD options for Half Disconnected telephones with OHAS enabled.

1 **HDOPT = 0** is the existing treatment without the OHAS Half Disconnect Enhancement. It is the default option and disables the Off-Hook Alarm Security Half Disconnect feature. Line Lockout treatment occurs after the normal Half Disconnect timer expires and Half Disconnect state is recognized.

2 **HDOPT = 1-10** indicates the maximum number of OHAS HD treatments given to the half disconnected analog (500/2500) type telephone. This option allows a limited number of OHAS HD treatments. If the telephone remains off-hook in the half-disconnect state after the maximum number of treatments has expired, Line Lockout occurs when the security DN disconnects after the last OHAS Half Disconnect treatment.

3 **HDOPT = CONT** provides a continuous application of the OHAS HD treatment, while the analog telephone remains in the half disconnected state. This option continues to call the security DN every time the HDTM expires until the analog (500/2500) type telephone goes on-hook.

Class of Service

To enable the OHAS HD feature the telephone must have CLS = Alarm Security Allowed (ASCA). Therefore when the HDTM timer expires, instead of giving the Line Lockout treatment, the OHAS HD treatment is given.

OHAS security DN

On a telephone basis an HDID is assigned. The HDID is the OHAS HD Index number. The values are 0 - 9. The Index number refers to the ten OHAS DNs you can program in the Customer Data Block. For example, if a telephone has HDID 1 assigned, OHAS HD treatment calls the security DN programmed for OHAS DN 1 (ODN1), in the Customer Data Block.

The OHAS HD Index can be configured to send calls to the same security DN as the existing OHAS Off-Hook Index (OHID) or a different security DN. This flexibility allows you to distinguish between regular OHAS dial tone/ interdigit time-out treatment calls (emergency situations) and OHAS HD treatments for half disconnect calls.

OHAS Half Disconnect Timer

With the OHAS Half Disconnect Enhancement feature enabled, the administrator can define the length of time before the OHAS HD treatment is given. The OHAS HD timer (HDTM) gives the average user enough time to complete the disconnect of the previous call by placing all the analog telephones on-hook. The length of the OHAS Half Disconnect timer can be defined from 1 to 600 seconds (10 minutes). The timer is programmable in one second increments. The HDTM starts after the half disconnect state is detected. The default for the HDTM is 30 seconds.

OHAS TTY record display

As with the existing OHAS feature, a message also prints out on the TTY terminal indicating the telephone which is receiving OHAS treatment. The message is the same for regular OHAS and OHAS Half Disconnect.

Each occurrence of an OHAS HD intercept treatment results in a message printout on the service change TTY or the active TTY. The content and the format of the OHAS HD message is the same as the regular OHAS off-hook or interdigit time-out message.

The following is an example of the record content:

```
OHAS000 2010 1 0 1 3 5:04:04  7/09/1998
```

The definitions of the fields are as follows:

OHAS000 = OHAS message indicator

2010 = DN (the DN of the analog (500/2500) type telephone receiving OHAS or OHAS Half Disconnect treatment)

1 0 1 3 = l s c u (the TN of the analog (500/2500) type telephone receiving OHAS or OHAS Half Disconnect Treatment)

Note: The TN for Small Systems and Succession 1000 systems is only two digits (c u).

5:04:04 = time stamp (when the OHAS or OHAS Half Disconnect Treatment is given)

7/09/1998 = date stamp

Operating parameters

While an analog (500/2500-type) telephone is in the half disconnect/Line Lockout state, the OHAS feature for emergencies cannot be triggered. OHAS will not work until the off-hook 500/2500 telephone goes on hook to disconnect the previous connection.

When OHAS Half Disconnect occurs, new calls cannot be initiated from the half-disconnected telephones.

If Party A goes on-hook at any time, the OHAS Half Disconnect treatment is canceled, since the disconnect is completed.

The OHAS Half Disconnect Timer is separate from the existing OHAS timer.

Digital telephones do not go into the half disconnect state. Digital telephones cannot share a TN with other telephones.

The feature does not apply to digital telephones since the half disconnect state does not apply to them.

The OHAS HD treatment is not provided for Attendant Consoles.

If the telephone remains off-hook in the half-disconnect state after the maximum number of OHAS HD treatments has expired, Line lockout occurs when the security DN disconnects after the last OHAS Half Disconnect treatment.

OHAS HD calls can be directed to a separate security DN to enable the user who answers the calls to distinguish between an Off Hook Alarm Security call and an Off Hook Alarm Security Half Disconnect Call.

Ringback tone can be heard at the off-hook analog telephone when the security DN is ringing. Anyone who uses one of the half-disconnected analog (500/2500-type) telephones can speak to the person who answers the security DN.

If Party A goes on-hook at any time, the OHAS Half Disconnect treatment is canceled, since the disconnect is completed.

If the connection is a trunk call and the far end does not disconnect completely, Party A will not go into the half disconnect state. The system treats Party B and Party A as if they are still on an active call.

The OHAS HD feature applies only to a single switch. It is not supported in a networking environment.

The OHAS HD security DN cannot be an Attendant DN.

The operation of the OHAS HD timer is impacted on systems with high traffic.

Feature interactions

Call Redirection

Call Redirection features defined for OHAS Half Disconnect security DN work as currently defined in the system. Call Redirection features include:

- Call Forward All Calls
- Call Forward No Answer
- Call Forward Busy
- Call Forward by Call Type
- Call Pickup
- Hunting

Conference

If an analog 500/2500 telephone user with the ASCA Class of Service is in a conference and all the other parties disconnect from the call while the user's telephone remains off hook, the OHAS Half Disconnect Enhancement feature applies to the half-disconnected telephone.

Line Lockout

If an analog telephone has the ASCA Class of Service, and it is in the half disconnected state, the OHAS HD treatment occurs if the customer-based OHAS Half disconnect option (HDOPT) is enabled. Choose HDOPT 1-10 or HDOPT = CONT. If HDOPT= 0 is selected, Line Lockout will occur.

If the telephone stays in the half disconnected state and the number of the OHAS HD treatments given to the telephone exceeds the maximum defined number, Line Lockout is given to the telephone after the last OHAS Half Disconnect treatment is given.

No Hold conference

The OHAS HD treatment works the same for a conference call initiated using No Hold Conference as for Conference.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Configure Off-Hook Alarm Security (OHAS) Directory Numbers (DNs), Half Disconnect treatment option, and the OHAS Half Disconnect timer.
2. LD 10 – Assign an ASCA Class of Service to the telephone. Associate the telephone with one of the ten Off-Hook Alarm Security Directory Numbers (ODN0-9) configured in LD 15.

   Note: The telephone is also programmed with an OHID, related to the OHAS feature.

LD 15 – Configure Off-Hook Alarm Security (OHAS) Directory Numbers (DNs), Half Disconnect treatment option, and the OHAS Half Disconnect timer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	OAS	Off-Hook Alarm Security (OHAS) options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
ODN0	xxxx	OHAS DN 0.
...		
ODN9	xxxx	OHAS DN 9.
ASTM	1 - (30) - 63	OHAS off-hook or interdigit timeout timer in seconds.
HDOPT		OHAS Half Disconnect treatment options:
	(0) 1-10 CONT	No OHAS HD treatment given. Maximum number of OHAS HD treatments. Continuous OHAS HD treatments.
HDTM	1- (30) - 600	OHAS Half Disconnect timer in seconds (in increments of 1 second).

LD 10 – Assign an ASCA Class of Service to the telephone. Associate the telephone with one of the ten Off-Hook Alarm Security Directory Numbers (ODN0-9) configured in LD 15.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	500/2500 telephones.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

CUST	xx	Customer number, as defined in LD 15
DES	d..d	Office Data Administration System Station Designator.
. . .	. . .	
DN	x...x	Directory Number.
....		
CLS	ASCA	Alarm Security Allowed. (ASCD) = Alarm Security Denied is the default.
...		
OHID	(0) - 9	OHAS ID index to OHAS security DN.
HDID	(0) - 9	OHAS Half Disconnect Index to OHAS HD security DN.

Feature operation

No specific operating procedures are required to use this feature.

Off-Premise Extension

Contents

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Feature operation 180

Feature description

The Off-Premise Extension (OPX) feature allows a single-line telephone serving as an extension to be located away from the customer premises. The loop limit is 1400 ohms to the station or equivalent long-line circuit interface. Distance varies depending on the gauge of wire used.

Refer to Nortel Networks technical publication *Circuit Card: Description and Installation* (553-3001-211) for additional information.

Operating parameters

The Off-Premise Extension (OPX) feature applies only to single line telephones. A QPC192 line circuit pack must be equipped.

Feature interactions

Refer to *Circuit Card: Description and Installation* (553-3001-211) for feature interactions.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 10 – Add or change Off-Premise Extension Class of Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(ONP) OPX	Telephone is an on-premises or off-premises extension.

Feature operation

There are no specific procedures required to operate this feature.

Off-Premise Station Analog Line Card

Contents

This section contains information on the following topics:

Feature description 181
Operating parameters 182
Feature interactions 183
Feature packaging 183
Feature implementation 183
Feature operation 189

Feature description

The Eight-port Off-Premise Station (XOPS) analog line card (NT1R20) is specific to North America and China as part of the Global Line Card program.

The XOPS card supports the current portfolio of peripheral equipment, and is designed for use in Off-premises Station (OPS) environments, connected through a Central Office (CO)/Public Exchange. It is also suited for campus system environments. Each of the units on the card can be configured to be operated as an OPS extension or in an On-premises (ONS) configuration.

The XOPS card requires a set of downloadable parameters for Termination and Balance Impedance values. These parameters are downloaded to the card whenever it is initialized or enabled. In addition, the analog cards require the loss/levels to be set for each unit on the card using the B34 Flexible Level message interface. ONS units receive loss/levels statically on Initialize or Enable.

Operating parameters

The XOPS card requires a Main Distribution Frame (MDF) wiring installation plan similar to trunks, rather than other line cards. Therefore, it will not be possible to interchange the XOPS card with another line card without rewiring the connections, or adjusting the Terminal Numbers (TNs) using service change.

The Classes of Service have been renamed to be consistent with industry standard terminology as follows: OPX is now called OPS; and ONP is now called ONS.

The jumper settings must be set in accordance with OPS and ONS Classes of Service.

The XOPS hardware will support Answer Supervision through Battery Reversal or Flash Hook.

No software support is provided for any Loopback from Extended Network Card (XNET) or XPEC to the XOPS line card.

The new XOPS line card uses B34 CODEC and Enhanced Extended Universal Trunk Card (EXUT) trunk circuitry. Therefore, the downloadable Termination Impedance (TIMP)/Balance Impedance (BIMP) combination parameter set, as defined for IPE EXUT, is likewise defined for the XOPS. The usage of TIMP/BIMP implies a limited number of downloadable combinations.

The XOPS is designed to work in North America using dynamic pad switching based on OPS and ONS Classes of Service. The card functions in a Static Loss Plan Download environment, but only the static levels associated with Analog Line Unit Short (ALUS) and Analog Line Unit Long (ALUL) are supported. In these situations, only Class of Service Long Line (LOL) or Short Line (SHL) has any meaning; OPS/ONS Class of Service of the unit is ignored.

As with the existing design, parameter download is not performed as part of enabling a Superloop, but is done as part of an initialization, or enabling of a unit, card, or peripheral shelf.

Hardware is compatible with the SL-100 circuit switched network, but software support for the SL-100 is not included as part of the XOPS feature.

Feature interactions

Due to the Loss Planning requirements for the XOPS card, the Global Line Card feature interacts with other Loss Planning features. The XOPS card must be able to operate in system environments that are using North American Transmission Plan, Static Loss Plan Download (SLPD), or Dynamic Loss Switching (DLS).

Feature packaging

Meridian 1 Superloop (XPE) package 203 is required, because the XOPS card can only operate in an IPE environment.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – At the TN prompt configure an XOPS card as a Double Density card on a Superloop.

2 LD 10 – The commands for creating or modifying an analog (500/2500 type) telephone type logical card block are modified to support the new card density for the XOPS card.

3 LD 10 – Use the "Easy Change" option to change only the BIMP and/or TIMP value, or the card density.

4 LD 10 – Additional checking is added to support MOV commands on XOPS units.

5 LD 10 – Additional checks are added to support CPY (copy) commands involving XOPS units.

6 LD 25 – Move card TNs from Superloop to Superloop.

7 LD 25 – Move card TNs from non-Superloop to Superloop.

8 LD 97 – Install or customize Static Loss Plan Download table.

9 LD 97 – Install or customize a Dynamic Loss Switching Alternate Levels table.

LD 10 – At the TN prompt configure an XOPS card as a Double Density card on a Superloop.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	 500 500M	Analog (500/2500 type) telephone data block. For Large Systems For Small Systems and Succession 1000 systems
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD DD 4D	Single, Double, or Quad Density.
DES	dddddd	1-6 alphanumeric character Office Data Administration System (ODAS) Station Designator.
...		
CLS	(OPS) (ONS) (LOL) (SHL)	Classes of Service ONS and OPS are supported. OPS is the default if the TN is on XOPS, otherwise ONS is the default. Classes of Service LOL and SHL are supported, but are not used for North America Loss Plan handling. LOL is the default if the TN is XOPS, otherwise SHL is the default.
...		
TIMP	(600) 900	Termination Impedance for XOPS unit. Prompted only if the specified TN is to be configured on an XOPS card (Double Density card on a Superloop).
BIMP	(3CM2) (600) 3COM 900	Balance Impedance for XOPS unit. 3CM2 is the default if the CLS is OPS, otherwise the default is 600.

LD 10 – The commands for creating or modifying an analog (500/2500 type) telephone type logical card block are modified to support the new card density for the XOPS card.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	CARDSLT	Card block for single line terminations.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD DD 4D	Single, double, or quad density.

LD 10 – Use the "Easy Change" option to change only the BIMP and/or TIMP value, or the card density.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Analog (500/2500 type) telephone data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
ECHG	(NO) YES	(Deny) allow the Easy Change option.

ITEM	TIMP ttt, BIMP bbbb CDEN cc CLS sss sss	Prompted only if the response to ECHG is yes. New ITEM responses TIMP or BIMP have been added, with the associated responses for each item (ECHG of TIMP and BIMP are only allowable for a Double Density card on a Superloop (XOPS)). TIMP: ttt is 600 or 900. BIMP is prompted next. BIMP: bbbb is 3CM2, 600, 900 or 3COM (BIMP should be set to 600 if the unit is configured with ONS Class of Service). ITEM is prompted next. CDEN cc is SD, DD or 4D (ECHG of CDEN continues to be supported, but existing code ensures that a single density card with at least one unit with Class of Service OPS cannot be changed to any other density. If CLS is changed to OPS, ONS, LOL, or SHL, TIMP is prompted next. Otherwise ITEM is prompted next.
TIMP	tttt	Prompted only if the response to ITEM was CLS of OPS, ONS, LOL, or SHL, and if CLS was changed from its previous setting. tttt is 600 or 900.
BIMP	bbbb	Prompted only if the response to ITEM is TIMP ttt or on change of CLS sss (bbbb is 3CM2, 600, 900, or 3COM).
ITEM	<CR>	Used to exit the ITEM prompt loop.

LD 10 – Additional checking is added to support MOV commands on XOPS units.

Prompt	**Response**	**Description**
REQ:	MOV	Move.
TYPE:	500	Analog (500/2500 type) telephone data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
TOTN	 l s c u c u	Destination Terminal Number For Large Systems For Small Systems and Succession 1000 systems

LD 10 – Additional checks are added to support CPY (copy) commands involving XOPS units.

Prompt	Response	Description
REQ:	CPY xx	Copy.
TYPE:	500	Analog (500/2500 type) telephone data block.
...		
CFTN		Copy From Terminal Number, prompted if REQ = CPY
	l s c u	For Large Systems
	c u	For Small Systems and Succession 1000 systems
SFMT	AUTO DN etc.	For AUTO and DN format types; the TNs are provided by the system.

LD 25 – Move card TNs from Superloop to Superloop.

Prompt	Response	Description
REQ	MOV	Move.
CUST	xx	Customer number, as defined in LD 15.
SRCL	0-156	Source loop.
DSTL	0-156	Destination loop.
MVSG	(NO) YES	Move segment.
SCHD		If attempting to move a Quad Density or Octal Density card on a Superloop to an XOPS card, or vice versa, an SCH6400 error message will be issued.
	l s c u TO l s c u	For Large Systems
	c u TO c u	For Small Systems and Succession 1000 systems

LD 25 – Move card TNs from non-Superloop to Superloop.

Prompt	Response	Description
REQ	MOV	Move.
CUST	<CR>	Customer number.
SRCL	0-156	Source loop.
DSTL	0-156	Destination loop.
MVSG	(NO) YES	Move segment.
SCHD	 l s c u TO l s c u c u TO c u	If attempting to move a Single Density, Double Density, or Quad Density card on a Superloop to an XOPS card, an SCH6400 error message will be issued. For Large Systems For Small Systems and Succession 1000 systems

LD 97 – Install or customize Static Loss Plan Download table.

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	LOSP XCTP XPE SUPL XNPD SYSP	Install or change the system Loss Plan.
TTYP	STAT	Modify the system SLPD table.
NATP	YES NO	North American Transmission Plan.
STYP	 PRED CSTM DISL	Static Loss Plan Download table type, where: PRED = Predefined table, CSTM = Customized table. DISL = Disable current active table If the response is PRED, TNUM is prompted. If CSTM is selected, SLPD port types are prompted after password verification. If response DISL is selected, SLPD will be disabled after password verification. If <CR> is entered, the table type is not changed (previously <CR> was treated as PRED).

TNUM	nn	SLDP Table number. nn is 1 to 25 Prompted if PRED is selected (REQ is prompted next).
PWD2	pppp ppp...p	Prompted only if STYP is CSTM and LAPW is restricted or the user logged in with the PWD1 password.
COTS	Rx Tx	CO trunk with SHL CLS.

LD 97 – Install or customize a Dynamic Loss Switching Alternate Levels table.

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	LOSP XCTP XPE SUPL XNPD SYSP	Install or change the system Loss Plan.
NATP	YES NO	North American Transmission Plan.
TTYP	DYNM	Modify the system DLS Alternate Levels table.
DTYP	PRED CSTM DISL	DLS Alternate Levels table type. If the response is PRED, TNUM is prompted. If CSTM is selected, DSL port types are prompted after password verification. If the response DISL is selected, DLS will be disabled after password verification. If <CR> is entered, the table type is not changed (previously <CR> was treated as PRED).
TNUM	nn	DLS Alternate Levels table number. nn is 1 to 3. Prompted if PRED is selected (REQ is prompted next).
PWD2	ppp ppp...p	Prompted only if DTYP is CSTM and LAPW is restricted or the user logged in with the PWD1 password.
COTS	Rx Tx	CO trunk with SHL CLS.

Feature operation

No specific operating procedures are required to use this feature.

On Hold on Loudspeaker

Contents

This section contains information on the following topics:

Feature description 191
Operating parameters 191
Feature interactions 192
Feature packaging 194
Feature implementation 195
Feature operation 197

Feature description

The On Hold on Loudspeaker (OHOL) feature is designed for brokers (dealers), and requires proprietary hardware to make use of its functionality. This feature provides brokers with the capability to monitor stock markets, while talking to one or several customers using the handset.

At any time the user can enter the call being monitored on the loudspeaker. This can also be done for the speech monitor unit either publicly by using the built in microphone (if provided) and the conversation will be heard on the channel, or privately by taking the call on the handset. Speech monitors work as loudspeakers, but with up to eight channels.

Operating parameters

This feature requires either proprietary loudspeakers that connect to M2616 sets, or a speech monitor system, and speech monitor units to work properly.

This feature is dependent upon access to conference cards and therefore each proprietary loudspeaker/speech monitor should have a conference loop assigned. Since the conference loops are used by the entire system, an option to separate normal conference traffic from “Dealer Group Traffic” is introduced.

One conference loop per system can be assigned as a Spare Dealer Conference loop. This loop is used as a backup if the conference loop assigned to an OHOL unit is in invalid state. This loop can only be used by the OHOL feature.

Feature interactions

Attendant Barge-in
Attendant Break-in
Attendant Busy Verify
Override

It will not be possible to Break-in/Barge-in/Busy Verify/Override into a call on loudspeaker as it is effectively on hold at the set.

Audible Reminder of Held Call

This feature works with the OHOL feature as for normal calls on hold (that is, it gives a reminder there are calls on hold). Therefore, it is not recommended to use this feature with the OHOL feature.

Call Forward All Types

No type of call forward can be activated on a set with Speaker Allowed Class of Service.

Call Transfer
Conference

It will not be possible to transfer or conference the loudspeaker call to another party.

Call Waiting
Camp-on
Ring Again

These features can be applied to a busy loudspeaker DN.

Conference Loops

The configuration of conference loops has been modified to indicate whether a conference loop is a Dealer or an ordinary conference loop.

Dial Access to Group Call

If a group call is initiated from a set with Dealer Allowed (Class of Service), the conference is built up on the assigned loop of the loudspeaker or speech monitor system channel since this is a potential OHOL call.

Group Hunt

Group Hunt to a loudspeaker DN can be programmed, but will be ignored if configured as Make Set Busy (MSB) by call processing.

Group Hunt

Group Hunt to a loudspeaker DN can be programmed, but will be ignored if configured as Make Set Busy (MSB) by call processing.

Held Call Clearing

Going on-hook when Held Call Clearing is activated will clear the loudspeaker as for a normal held call. Therefore, it is recommended not to use this feature with the OHOL feature.

Hold

The feature is limited to use with normal hold or automatic hold. Deluxe hold will be ignored by call processing.

Hot Line
Voice Call

It is possible to program these keys with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Hot Line Two Way

This feature can be used with the speech monitor system. The DN of the speech monitor system channel is configured as the DN for the HOT line key.

Hunting
Call Forward

Hunt/Call Forward to a loudspeaker DN can be programmed, but will receive intercept treatment as for direct dial to the loudspeaker DN.

Music

If Music on Hold is equipped it will not be heard by either party during a loudspeaker call.

Ring Hold LED Status

This feature reverses the lamp indication of ringing and held calls. With this feature activated, held calls will fast flash and ringing calls will slow flash.

Single Call Ringing

If a single call ringing loudspeaker DN (an analog [500/2500 type] telephone with CLS = SPKA) is dialed, intercept treatment is provided.

Telephones - Analog (500/2500 type)

The loudspeaker and speech monitor system channels are configured as analog (500/2500-type) sets with Speaker Allowed Class of Service (CLS = SPKA). These sets are in a permanent off-hook state. The units are recognized as in lockout state by the system.

Feature packaging

On Hold on Loudspeaker (OHOL) package 196 is required to operate this feature.

It is recommended to have the Autohold feature configured with this feature to simplify its operation.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Assign Dealer Conference loop and Spare Dealer Conference loop.

2 LD 10 – A new Class of Service is added to this overlay to allow an analog (500/2500 type) telephone to be assigned as a loudspeaker DN. A new prompt, DCLP (Dealer Conference Loop), has been added to configure the assigned conference loop.

3 LD 11 – Configure the M2616 set with LSPK key. Only one key can be configured per set.

4 LD 11 – Configure a set with a DN key corresponding to a speech monitor system channel.

LD 17 – Assign Dealer Conference loop and Spare Dealer Conference loop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Common equipment parameters.
...		
- CONF	0-158	Conference loops.
	D0-D158	Conference loop number assigned as Dealer Conference loop.
	S0-S158	Conference loop assigned as Spare Dealer Conference loop. It is strongly recommended that this loop is in the same group as the unit planning to use this loop to minimize the use of intergroup timeslots.
	X0-X158	To remove entry.

LD 10 – A new Class of Service is added to this overlay to allow an analog (500/2500 type) telephone to be assigned as a loudspeaker DN. A new prompt, DCLP (Dealer Conference Loop), has been added to configure the assigned conference loop.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	500	Analog (500/2500 type) telephone.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	SPKA	Speaker allowed.
DCLP	xx	Assign loop number with or without option Dealer Conference loop.

LD 11 – Configure the M2616 set with LSPK key. Only one key can be configured per set.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	M2616	Meridian Modular set.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	DELA	Dealer allowed.
KEY	xx LSPK nnnnnn	Loudspeaker, where xx is the key number, and nnnnnn is the LSPK DN which is the same DN as for the OHOL unit.

LD 11 – Configure a set with a DN key corresponding to a speech monitor system channel.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	M2616	Meridian Modular set.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	DELA	Dealer allowed.
KEY	xx SCR nnnnnn	xx is the key number. nnnnnn is DN which is the same DN as for the speech monitor system channel. When this DN is put on hold, the speech monitor unit will automatically be switched on.

Feature operation

Proprietary Loudspeaker System

This system consists of a M2616 set with a Loudspeaker (LSPK) key configured and an attached add-on module which has been modified to work as a loudspeaker. The proprietary loudspeaker is to be used when a user needs to be able to monitor one call on the loudspeaker at the same time as monitoring another call on the handset.

The loudspeaker is connected to a 500 line card and is in a permanent off-hook state. The DN of the loudspeaker must be Single Call Ringing (SCR).

Sets with this configuration are allowed to manually put calls onto the loudspeaker. The call to be put onto the loudspeaker has to be on hold at the set.

To activate the loudspeaker, press the LSPK key and then press any DN key on hold. The held call is put onto the loudspeaker and will be heard publicly. A user can enter into the call by using the handset on the loudspeaker (if provided). While the loudspeaker is active, any other call will be maintained on the handset. More than one call can be put on hold on the set, however only one call at a time can be switched to the loudspeaker.

To release the call from the loudspeaker, the active call on the handset has to be put on hold (either by automatic hold or manual hold) or released.

Attempts to activate a call onto the loudspeaker when busy will be ignored.

Speech Monitor System

The speech monitor system is used in an environment where several users need to listen to the same call publicly. The speech monitor system enables calls to be automatically extended to a loudspeaker. The loudspeaker in this scenario is the speech monitor unit.

The speech monitor unit has a number of speech monitor system channels (a maximum of eight) available. These channels can be switched onto the speech monitor unit and heard publicly. Each speech monitor system channel has a SCR DN configured. This SCR DN has a mixed appearance on a key (DN or HOT) on a user's set. Several users can have the same mixed DN on their set (Multiple Appearance SCR DN). The set can also have a two-way HOT line key with the same DN as a speech monitor system channel. While monitoring up to eight calls on the speech monitor unit, the users' handsets are free to maintain other calls.

The speech monitor system channels are attached to a 500 line card which is in a permanently off-hook state. The unit is recognized as in lockout state by the system.

The speech monitor system channel can be activated from DN keys or two-way HOT line keys where the DN for the HOT line is a mixed appearance with a DN of a speech monitor system channel. The user makes a call from this specific DN or HOT line key. When the call is established the user then puts the call on hold by using automatic hold or manual hold. The corresponding channel on the speech monitor system will automatically be activated. The call can then be heard on the speech monitor unit when the channel is selected. At any time the user can enter the call on the speech monitor unit by using the built-in microphone (if provided) and this two-way conversation will be heard on the loudspeaker in addition to any other channels active on the loudspeaker.

To talk privately on one of the calls being monitored on the speech monitor unit, the user takes the call on the handset of the phone. This conversation will not be heard on the loudspeaker, but any other user with the same DN appearance will be able to enter the call by going off-hook and establishing a multiple appearance conference.

If the user presses the Release key while active on a call that appears on a speech monitor system channel, the call is disconnected from all DN appearances, including the speech monitor system channel.

It is not possible to prevent the speech monitor unit from becoming active. If a user no longer wishes to listen to the speech monitor, the unit needs to be switched off manually.

On-Hook Dialing

Contents

This section contains information on the following topics:

Feature description 201
Operating parameters 201
Feature interactions 202
Feature packaging 202
Feature implementation 202
Feature operation 202

Feature description

The On-Hook Dialing feature enables a Meridian 1 proprietary telephone user to make a call without lifting the handset. Signaling tones and the voice of the called party are heard over the loudspeaker. For two-way communication, the user must lift the handset or activate the Handsfree unit if equipped.

Operating parameters

The On-Hook Dialing feature does not apply to analog (500/2500 type) telephones.

Feature interactions

LOGIVOX Telephone

Because of the firmware on the LOGIVOX set, the DN key 0 is automatically selected when the first digit is dialed, and no other DN has been selected.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Optional Outpulsing Delay

Contents

This section contains information on the following topics:

Feature description 203
Operating parameters 203
Feature interactions 203
Feature packaging 204
Feature implementation 204
Feature operation 204

Feature description

The Optional Outpulsing Delay (OOD) feature increases to three seconds the Start of Dialing Delay used for automated dialing on loop start Central Office (CO) trunks. This feature is required for system connection in some countries.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Features that automatically dial digits onto a loop start CO trunk are provided with an additional delay. These features include the following:

- Stored Number Redial

- Autodial
- Speed Call
- Call Forward All Calls
- Basic Alternate Route Selection/Network Alternate Route Selection (BARS/NARS)
- System Speed Call, System
- Network Speed Call, and
- Flexible Hot Line.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Outgoing Hold Timer Increase

Contents

This section contains information on the following topics:

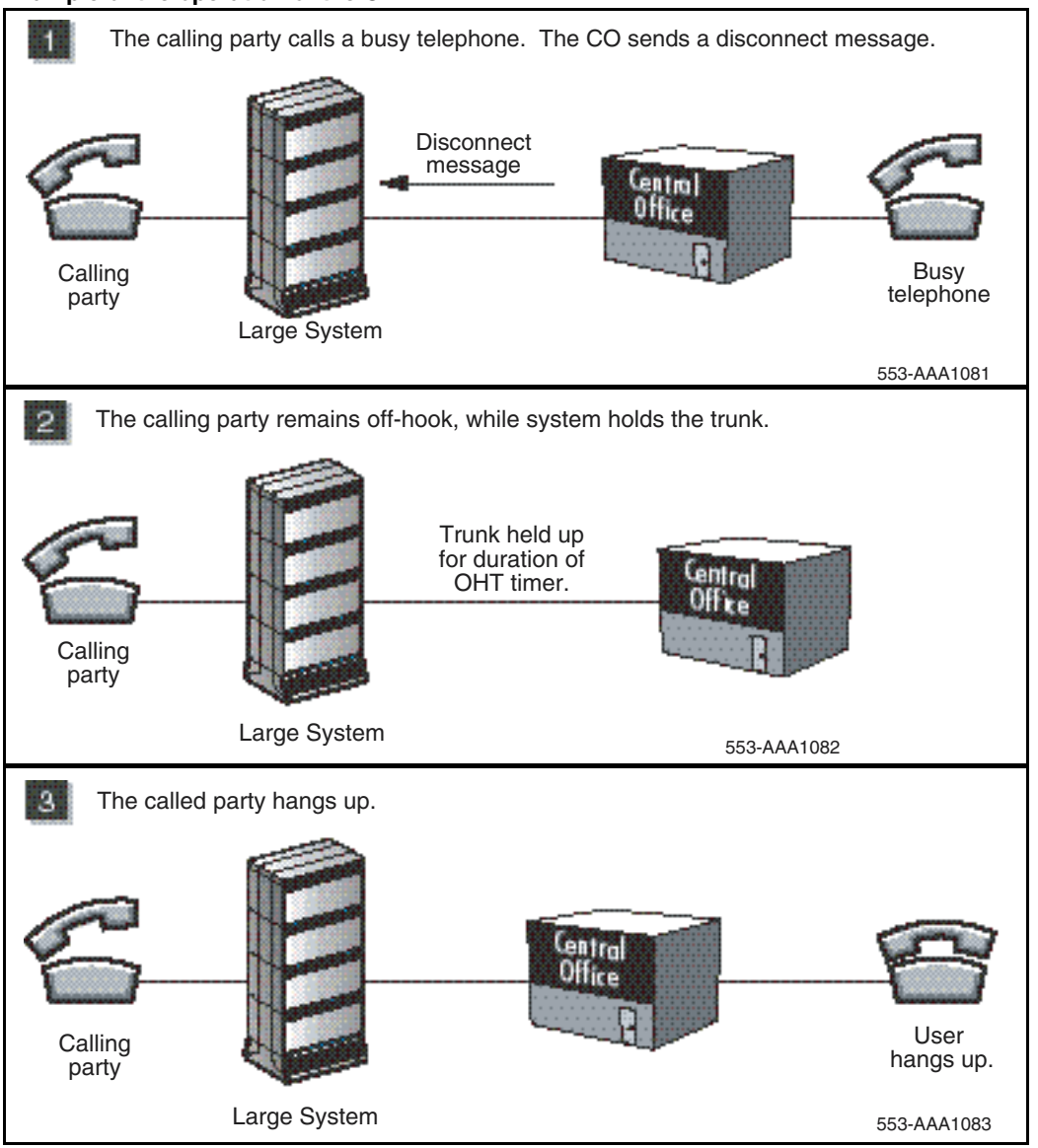
Feature description 205
Operating parameters 207
Feature interactions 208
Feature packaging 208
Feature implementation 208
Feature operation 209

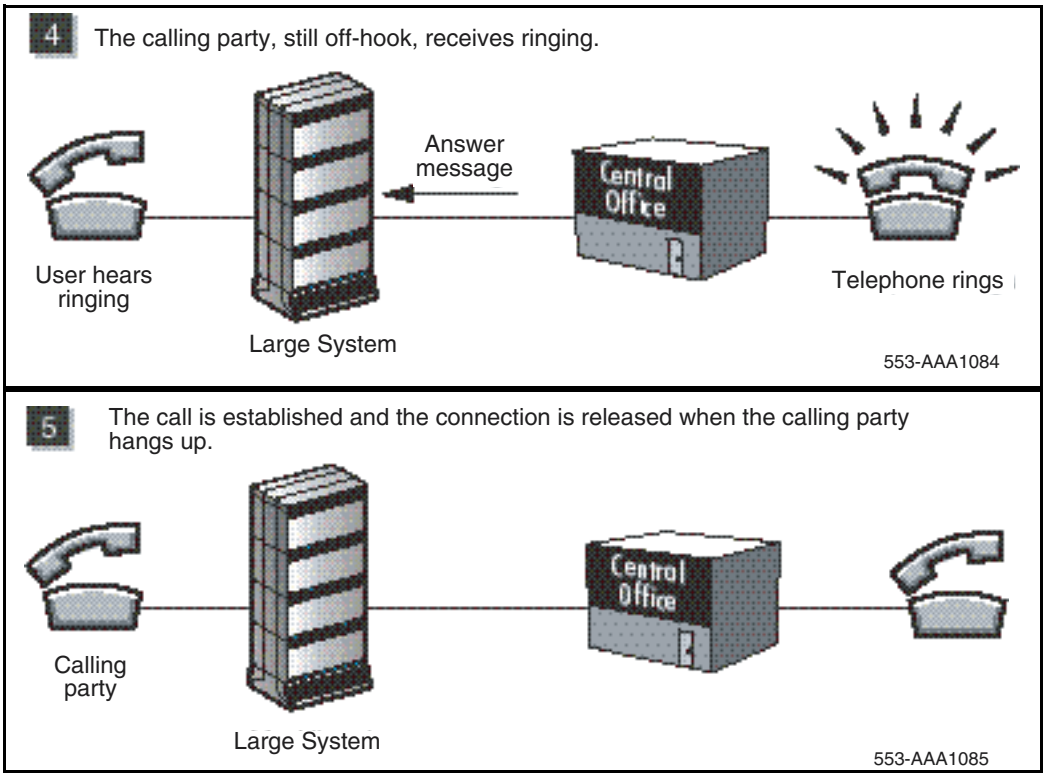
Feature description

The increase to the Outgoing Hold Timer (OHT), included in the Operator Call Back feature (OPCB), increases the time the system holds a trunk after it receives a disconnect message from a Central Office. The OHT applies to situations where Calling Party Control is active.

Figure 5 on page 206 shows an example of a Calling Party Control (CGPC) call, where the calling party controls the disconnect.

Figure 5
Example of the operation of the OHT





On an outgoing call, a CO can send a disconnect message back to the system during call establishment. The system does not disconnect the outgoing call until the OHT has expired. If the CO sends an answer message to the system before the timer expires, the originator is connected to the called party.

The OHT determines the length of time the system holds a trunk after receiving a disconnect message. The maximum is 126 seconds. The timer is programmed in increments of 2 seconds. The default value is 30 seconds.

Operating parameters

This feature enhances the existing OHT capability provided by Package 126.

The OPCB OHT is available on analog and DTI2 trunk interfaces. It is not supported on DTI 1.5 trunk routes.

Feature interactions

Outgoing Hold Toll Timer

When the CO sends a disconnect message on an outgoing toll call, the Outgoing Hold Toll Timer (OHTT) disconnects after a maximum of 90 seconds. The OHTT can be programmed in increments of two seconds.

Feature packaging

This feature requires Operator Call Back (OPCB) package 126.

Feature implementation

LD 16 – Configure the OHT on the trunk route, at the OHFT prompt.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	RDB	Route Data Block.
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
CNTL	YES	Changes to control or timers.
...		
OPCB	YES	Enable the Operator Call Back feature.
...		
OHT	0-(30)-126	Outgoing Hold Timer in seconds (programmed in increments of two seconds).
. . .		

Feature operation

No specific operating procedures are required to use this feature.

Out-of-Service Unit (OOSU)

Contents

This section contains information on the following topics:

Feature description 211
Operating parameters 211
Feature interactions 212
Feature packaging 212
Feature implementation 212
Feature operation 213

Feature description

The ability to mark a unit as “Out-of-Service” is a feature that is part of the Global Line Cards program. This capability is accomplished through Service Change. A unit marked Out of Service cannot be configured as any other type of unit without first removing it from the Out-of-Service state. A unit marked Out of Service stays Out of Service through Initialization or SYSLOAD operation. This feature reduces the number of cards that must be replaced in situations where only one, or a few circuits, fails to work in the field. In addition, the capability enables support personnel to change high density cards at convenient low-traffic periods.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 10 – At the TYPE prompt respond with OOSLT to designate single-line terminal units as Out of Service.
2. LD 11 – At the TYPE prompt, respond with OOSMLT to designate multi-line terminal units Out of Service. The capability to make any unit Out of Service, regardless of the card type or density, is also designated by this response.

LD 10 – At the TYPE prompt respond with OOSLT to designate single-line terminal units as Out of Service.

The OOSLT response provides the capability to designate an analog (500/2500 type) telephone as being Out of Service, regardless of card density. This Out-of-Service status survives a SYSLOAD. To reconfigure a unit as another type of unit it is necessary to first remove the unit from its Out-of-Service status, and then reconfigure it as NEW.

Prompt	Response	Description
REQ:	NEW OUT	New, or remove.
TYPE:	OOSSLT	Out-of-service single-line terminal unit.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems If the REQ is NEW, a check is made to verify that the card already exists, and the unit specified is not already configured. If the REQ is OUT, a check is made to verify that the unit is marked Out of Service. If the unit specified to be removed is the last configured unit on the card, the card blocks associated with the logical card are removed.

LD 11 – At the TYPE prompt, respond with OOSMLT to designate multi-line terminal units Out of Service. The capability to make any unit Out of Service, regardless of the card type or density, is also designated by this response.

Prompt	**Response**	**Description**
REQ:	NEW OUT	New, or remove.
TYPE:	OOSMLT	Out of Service multi-line terminal unit.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems If the REQ is NEW, a check is made to verify that the card already exists, and the unit specified is not already configured. If the REQ is OUT, a check is made to verify that the unit is Out of Service. If the unit specified to be removed is the last configured unit on the card, the card blocks associated with the logical card are removed.

Feature operation

No specific operating procedures are required to use this feature.

Overlay 45 Limited Repeats

Contents

This section contains information on the following topics:

Feature description 215
Operating parameters 216
Feature interactions 216
Feature packaging 216
Feature implementation 216
Feature operation 217

Feature description

Overlay 45, the Background Continuity Diagnostics, is automatically loaded whenever a power fault is detected, and runs in the background. This feature allows a limit to be placed on the number of times that background continuity tests are run by this overlay. This limit is system configured in LD 17, and may have a value from 0-31. Once the defined value has been reached, the regular background programs are restored. The alarm is not cleared. Since the alarm has not been cleared, overlay 45 is not reloaded before the end of the current midnight routine cycle. At the end of the midnight cycle, the alarm is cleared by the overlay supervisor.

If there are no midnight routines, overlay 45 starts a timer which is decreased at regular intervals by the work scheduler. The alarm is not cleared at this point. Therefore, overlay 45 is not reloaded for an alarm condition. When the timer expires, the work scheduler clears the alarm. If another alarm condition arises, overlay 45 is automatically loaded and runs as described above.

Operating parameters

The system overlay loader checks for power alarms and sets the relevant task request bit, if found. This overlay loader is modified to ignore power alarms once the limit defined for the overlay repeats has been reached, until the end of the current midnight routine cycle, if there is one.

It is advised that a printer be used to obtain hard copy information on the continuity tests run by overlay 45, rather than relying on the history file, if one is available.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP) package 131.

Feature implementation

LD 17 – Configure Overlay 45 Limited Repeats parameters, at the CY45 prompt.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CFN OVLY	Configuration Record. Overlay area options.
...		
- CY45	(0)-31	Cycles of LD 45. Cycles of LD 45 can be run whenever a fault is detected. If any number from 1 to 31 is entered, that is the number of times LD 45 will run under fault conditions. If 0 is entered, the system will perform as before without limiting the number of LD 45 runs.

Feature operation

No specific operating procedures are required to use this feature.

Overlay Cache Memory

Contents

This section contains information on the following topics:

Feature description 219
Operating parameters 220
Feature interactions 221
Feature packaging 221
Feature implementation 222
Feature operation 222

Feature description

The Overlay Cache Memory feature uses Protected Data Storage (PDS) as a cache area for storing overlays loaded from disk. The cache memory overlays are accessed much faster than those on disk, reducing the load time to approximately one second.

A maximum of 32 overlays can reside in Overlay Cache Memory at one time. The CACH prompt in option 17 defines the number of cache memory buffers allocated in protected memory. Each overlay resides in a buffer. A zero entry deactivates this feature and requires all overlays to be loaded from disk.

Each buffer requires 19K of Protected Data Storage (PDS). If there is insufficient memory to store the number of buffers requested, a warning message follows the option 17 prompt sequence. The message indicates that more memory is required to store all the caches requested.

If a small number of cache memory buffers are allocated, frequently used overlays may be removed from protected memory by seldom used overlays. The PRTY prompt in option 17 sets an overlay priority flag. A priority flag prevents the removal of an overlay from cache memory. The number of priority flags set cannot exceed the number of cache memory buffers specified.

When an LD xx command is entered, the cache memory is checked for the requested overlay. If the requested overlay is in cache memory, its data portion is rapidly copied to the regular overlay area.

A requested overlay that is not in cache memory is loaded from the disk into the normal overlay area and simultaneously stored into a cache memory buffer, if one is available. If one is not available, the new overlay overwrites another in the cache memory.

If an overlay is loaded from disk and no unused buffer area exists, the overlay used longest ago without its priority flag set is removed and replaced by the new overlay.

Operating parameters

If the feature is deactivated with a zero (0) entry at the CACH prompt in LD 17, no cache memory exists and all overlays are loaded from disk.

Cache memory is not affected by a system initialization. After a system initialization, it is not necessary to reload overlays from the disk.

Each buffer requires 19K of PDS. The number of cache memory buffers allocated to the system is limited by the availability of spare memory. If enough memory exists, a maximum of 32 cache memory buffers is allowed. Each buffer stores one overlay.

The number of overlay priorities (PRTY) that can be set is dependent upon software version.

To load an overlay from disk, use the command LD xx D. This is necessary for the system to determine which overlay to read. The LD xx D command loads the overlay from disk and overwrites the same overlay existing in cache memory.

Using the LD xx D command to force load an overlay from disk does not simultaneously support the peripheral download SUSP command.

When overlays are stored in cache memory, the ENLT and DIST commands are not supported.

The system automatically stores and retrieves overlays from cache memory. If the cache area is full when a new overlay is requested, the overlay gone unused the longest without a priority flag set is removed and replaced by the new overlay. Daily routines and background-loaded overlays are not stored in cache memory.

Conversion and upgrades

Due to memory requirements, installing a new issue of software or the same issue with additional features may reduce the number of cache buffers that can be allocated. A warning message indicates this reduction has occurred.

If this reduction causes the number of overlay priorities to exceed the maximum number of cache buffers, the overlay priorities are reduced to equal the number of cache buffers. The priorities are automatically reduced by beginning with the highest overlay number and working downward.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in the base system software.

Feature implementation

LD 17 – Enter the number of overlay buffers and priority of stored overlays.

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	CFN OVLY	Configuration Record. Overlay area options.
OVLY	YES	Change overlay area.
- CACH	(0) 2-32	Number of overlay buffers held in cache memory. Entering 0 disables the feature.
- PRTY	xx xx xx xx...	Set priority for the stored overlays. Priority can be set only for the number of overlays specified in CACH. xx = the overlay number. An X preceding the number deletes the priority flag for that overlay.

Feature operation

No specific operating procedures are required to use this feature.

Override

Contents

This section contains information on the following topics:

Feature description 223
Operating parameters 224
Feature interactions 224
Feature packaging 228
Feature implementation 228
Feature operation 230

Feature description

The Override feature provided in base system software allows a user to enter into an established connection. A warning tone notifies the talking parties that a third party is about to enter the conversation. The warning tone is an initial one-second burst, followed by a 256 millisecond burst repeated every 16 seconds. The Override feature can be used after a user has dialed a busy Directory Number (DN).

Forced Camp-On and Priority Override

The Forced Camp-On and Priority Override features provide enhancements to the basic Camp-On feature. Forced Camp-On is similar to the regular Station-to-Station Camp-On, except that it can be done without an internal or external call on hold. Forced Camp-On is activated automatically (if Automatic Forced Camp-On is defined); or it can be activated manually using the Enhanced Override (EOVR) key on system telephones or the Enhanced Override Flexible Feature Code on analog (500/2500 type) telephones.

The telephone performing the override must have a priority level equal to or higher than the telephone being overridden. Priority Override is activated by dialing the Override Flexible Feature Code on analog (500/2500 type) telephones, or by pressing the Override key (OVR) on system telephones.

Operating parameters

On Meridian 1 proprietary telephones, a separate Override key must be assigned. An associated lamp is not required.

On analog (500/2500 type) telephones, a Flexible Feature Code (FFC) is required to override a call.

Override cannot be used to enter an established connection if any party (telephone or trunk) has Warning Tone Denied Class of Service. In this case, overflow tone is heard.

The system must have a conference loop.

Feature interactions

Attendant Break-In

When one system telephone has overridden an existing call to establish a Conference call, Break-In is temporarily denied. The attendant is notified using the Override tone.

Telephones with a toll operator break-in call cannot be overridden. Overflow tone is returned to telephones attempting Priority Override.

Automatic Redial

An Automatic Redial (ARDL) call cannot be overridden. This is done to avoid creating a conference when a tone detector is involved.

Call Forward/Hunt Override Via Flexible Feature Code

It is possible to use Priority Override after using the Call Forward/Hunt Override FFC and encountering a busy set.

Call Party Name Display

When Overriding an established call, the displays of the other telephones show the DN and name of the overriding party.

Camp-On

Station-to-Station Camp-On and Attendant Camp-On are not affected by Forced Camp-On or Priority Override. The following new Classes of Service affect only Forced Camp-On:

- Camp-On From Another Telephone Allowed (CPFA)
- Camp-On From Another Telephone Denied (CPFD)
- Camp-On To Another Telephone Allowed (CPTA)
- Camp-On To Another Telephone Denied (CPTD)

The Station Camp-On (SCMP) package 121 is required to return busy tone instead of ringback tone to the party camping on.

Camp-On, Forced

When Priority Override is activated, it replaces normal override. Once Priority Override has been performed on a set, its Digit Display shows the DN of the overriding set.

Charge Account and Calling Party Number

When Charge Account is used during active Override, some digits may be lost. When entered with Override in conference, a Charge Account number is accepted and no digits are lost.

China – Attendant Monitor

A set may operate override to join into a desired call. If the desired call is being Attendant Monitored at the time, one of the following occurs:

- If the desired call is a conference call, the override attempt is blocked as per existing operation.
- If the call is a simple one with the Attendant Monitoring with no tone, the override attempt is successful and Attendant Monitor is deactivated.
- If the call is a simple one with the Attendant Monitoring with tone, the override attempt is blocked.

Conference

Override cannot be used to enter a Conference call.

Do Not Disturb

Telephones with Do Not Disturb enabled cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Group Hunt

Override will not be supported.

Hot Line

A Hot Line call can be entered using the Override feature.

Make Set Busy

Telephones with MSB active cannot be overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.Voice Call is blocked by MSB.

Multi-Party Operations

With Multi-Party Operations (MPO), when a consultation call is made on a set equipped with Priority Override, a control digit has to be dialed from the set to perform a recall and return the call on hold.

Network Intercom

An internal Hot Type I call never returns busy, unless the call became a non-Hot Line call due to the Hot Line key being busy. In this case, the call behaves like a normally dialed call, and override can be used upon receipt of a busy signal.

Night Restriction Classes of Service

If Priority Override and Night Restriction for Priority Override Class of Service (NROA) are assigned, Priority Override will be operational for the set only when Night Service is in effect.

On Hold on Loudspeaker

It will not be possible to Override into a call on loudspeaker as it is effectively on hold at the set.

Override, Enhanced

Priority Override

If Priority Override is equipped, it replaces Override when using the OVR key or OVRD FFC. However, Override can be simulated by using the default PLEV, 2, for all trunk routes and sets.

Periodic Camp-on Tone

The Periodic Camp-On Tone has precedence over Override intrusion tone.

Phantom Terminal Numbers
Call Forward

Call Forward cannot be overridden on phantom terminal numbers. The overflow tone occurs if an Override is attempted.

Ring Again

Ring Again is the only other feature currently available once a busy telephone has been encountered. Ring Again is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Uninterrupted Line Connections

Override cannot be applied to stations with a Warning Tone Denied Class of Service.

Feature packaging

Override is included in base system software.

For analog (500/2500 type) telephones, Flexible Feature Code (FFC) package 139 must be equipped.

Forced Camp-On/Priority Override (POVR) is package 186.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Allow Override for analog (500/2500 type) telephones.

2 LD 11 – Add or change Override for Meridian 1 proprietary telephones.

3 LD 14 – Define Warning Tone Allowed for trunks to permit Override.

4 LD 57 – Configure Flexible Feature Code (FFC) for Override on an analog (500/2500 type) telephones.

LD 10 – Allow Override for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

CLS	(OVDD) OVDA (XFD) XFA (WTA) WTD	Override (denied) allowed for this telephone. Transfer (denied) allowed. Warning Tone (allowed) denied (WTA is required to be overridden).

LD 11 – Add or change Override for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = Meridian 1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Warning Tone (allowed) denied (WTA is required to be overridden).
KEY	xx OVR	Override key (must be key 34 for the M3000).

LD 14 – Define Warning Tone Allowed for trunks to permit Override.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaa	Trunk type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Warning Tone (allowed) denied (WTA is required to be overridden).

LD 57 – Configure Flexible Feature Code (FFC) for Override on an analog (500/2500 type) telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.
CUST	xx	Customer number, as defined in LD 15
CODE	OVRD	Change Override access code.
OVRD	xxxx	Override access code.

Feature operation

To override a call in progress from a Meridian 1 proprietary telephone:

1 Dial the number. You hear a busy tone.

2 Press **Override**. Everyone hears a one-second tone burst.

3 You are connected to the call.

To cancel Override from a Meridian 1 proprietary telephone:

1 Press **Rls** or hang up.

2 You are disconnected. The original call remains active.

To override a call in progress from an analog (500/2500 type) telephone:

1 Dial the number. You hear busy tone.

2 Flash the switchhook or press **LINK**.

3 Dial the Flexible Feature Code (FFC) for Override. Everyone hears a one-second tone burst.

4 You are connected to the call.

To cancel Override from an analog (500/2500 type) telephone:

1 Press **Rls** or hang up.

2 You are disconnected. The original call remains active.

Override, Enhanced

Contents

This section contains information on the following topics:

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Operating parameters 237
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Feature packaging 241
Feature implementation 241
Feature operation 247

Feature description

The use of the Forced Camp-On and Priority Override features together results in Enhanced Override (EOVR).

Forced Camp-On

Forced Camp-On allows a call to be camped on and a warning to be given before the Priority Override operation. It differs from normal Camp-On in that both internal and external calls can be camped on, rather than just external calls as with the Camp-On feature. The Forced Camp-On may be automatic or manual. The manual operation requires the use of the Enhanced Override (EOVR) feature.

Forced Camp-On can be used as a feature by itself or in conjunction with Priority Override. The combination of the two features is referred to as Enhanced Override (EOVR).

For manual Forced Camp-On an analog (500/2500 type) telephone, the user has to dial the EOVR Flexible Feature Code (FFC), while a Meridian 1 proprietary telephone user has to use the EOVR key.

A second operation of the EOVR key or FFC executes Enhanced Override.

Forced Camp-On is similar to Station-to-Station Camp-On, except that Forced Camp-On can be done with either no call on hold or an external or internal call on hold. It can be done automatically or manually, which is determined by the response to the Automatic Forced Camp-On (AFCO) prompt in LD 15.

For manual operation, once a busy telephone has been reached, the first depression of the EOVR key or the first dialing of the EOVR FFC attempts Forced Camp-On. If successful, Forced Camp-On introduces Camp-On tone into the connection. If unsuccessful, overflow (fast busy) tone is returned to the party attempting the Forced Camp-On.

For Forced Camp-On to be allowed, all other methods of call termination must have been tried, and the last one must be Camp-On. If Station-to-Station Camp-On or Automatic Forced Camp-On has occurred, or Forced Camp-On has been excluded by the new telephone options, then the first depression of the EOVR key or first dialing of the EOVR FFC introduces Enhanced Override. If, however, Forced Camp-On is denied by existing Camp-On restrictions, Enhanced Override is also denied.

Priority Override

The Priority Override (POVR) feature allows users to break in to an established connection. To do this, analog (500/2500 type) telephone users use the OVRD Flexible Feature Code (FFC), and Meridian 1 proprietary telephone users use the Override (OVR) key before Camp-On.

The Priority Override Level (PLEV) restrictions apply to both Enhanced and Priority Override.

For Priority Override, the overriding telephone must have a Priority Override Level (PLEV) greater than or equal to the PLEV of the telephone or trunk to be overridden.

For an analog (500/2500 type) telephone, a recall followed by dialing the Priority Override FFC, (Override FFC with Priority Override package 186 equipped), breaks into the connection and establishes a conference between all three parties and sends an override tone. For a Meridian 1 proprietary telephone, the OVR key is used in place of the FFC.

In order for Priority Override to be allowed, all telephones and trunks involved must have Warning Tone Allowed (WTA) Class of Service. Each telephone and trunk route (TIE, DID, and COT) is assigned a PLEV as outlined in Table 4.

Table 4
PLEV assignments

PLEV	Indication
0	This telephone or route cannot be overridden; if assigned to a telephone, the telephone cannot override.
1	This telephone or route can be overridden; if assigned to a telephone, the telephone cannot override.
2	This telephone or route can be overridden by telephones assigned level 2 through level 7; if assigned to a telephone, the telephone can override level 1 and level 2.
3-6	(Similar to level 2) This telephone or route can be overridden by telephones assigned an equal or higher level; if assigned to a telephone, the telephone can override lower and equal levels, except level 0.
7	This telephone or route can be overridden by another level 7 telephone only; if assigned to a telephone, the telephone can override level 1 through level 7.

Several combinations of the Forced Camp-On and Priority Override are highlighted in the following list:

- Responding to the Automatic Forced Camp-On (AFCO) prompt in LD 15 with “NO,” configuring Meridian 1 proprietary telephones with only Override (OVR) keys, and defining the Override (OVRD) Flexible Feature Code (FFC) disallows the use of Forced Camp-On.

- Responding to the Automatic Forced Camp-On (AFCO) prompt in LD 15 with "NO" and setting the Priority Level (PLEV) to 0 and the Camp-On Classes of Service to Camp-On From Another Telephone Denied (CPFD) and Camp-On To Another Telephone Denied (CPTD) gives manual Camp-On only.
- Configuring the EOVR FFC for analog (500/2500 type) telephone users and equipping Meridian 1 proprietary telephones with EOVR keys gives the users the ability to use only Priority Override (using OVR key or OVRD FFC) or Forced Camp-On followed by Priority Override (pressing the EOVR key twice or using EOVR FFC).
- Responding to the Automatic Forced Camp-On (AFCO) prompt in LD 15 with "YES," configuring Meridian 1 proprietary telephones with only Override (OVR) keys, and defining the Override (OVRD) Flexible Feature Code (FFC) automatically applies Forced Camp-On in situations where it is allowed, and allows the use of the OVR key and FFC to implement Priority Override.
- Using an EOVR key or FFC with a response of "YES" to the AFCO prompt in LD 15 simulates the Override (OVR) key or FFC unless Forced Camp-On was denied initially, in which case the Forced Camp-On would be re-attempted.

The following table summarizes the various combinations:

Table 5
Summary of various combinations of Forced Camp-On and Priority Override.

	AFCO = NO	**AFCO = YES**
OVR FFC or key	Attempts Priority Override.	Attempts Priority Override whether Forced Camp-On occurred or not.
EOVR FFC or key	First use attempts Forced Camp-On, unless station is camped on, then Priority Override is attempted. Second use attempts Priority Override.	If automatic Forced Camp-On was denied, re-attempts Forced Camp-On; otherwise Priority Override is attempted.

Operating parameters

The Flexible Feature Codes (FFC) package 139 must be equipped for Forced Camp-On and Priority Override to be available from analog (500/2500 type) telephones.

For analog (500/2500 type) telephone activation, the Multi-Party Operations (MPO) package 141 must be equipped, with the "YES" as the response to the RALL prompt in LD 15 to ensure that register recalls are required before dialing control digits. The OVRD and EOVF FFCs defined must not start with the same digit as one of the control digits. The control digits are defined in LD 15 and are printed as part of the Customer Data Block in LD 21.

If Priority Override is equipped, it replaces Override when the user uses the OVR key or OVRD FFC. However, Override can be simulated by using the default value, 2, for all trunk routes and telephones.

Telephones or trunks involved in any of the following cannot be camped on or overridden:

- Non-established call
- Conference call
- Attendant call
- Attendant call using Centralized Attendant Service (CAS), Primary Rate Interface (PRI), or Integrated Services Digital Network (ISDN) trunk
- Make Set Busy
- Do Not Disturb
- Automatic Call Distribution (ACD) call
- Operator Call Back
- Hold
- Data call
- Release Link call, or
- Parked call.

Call Forward and Hunting take precedence over Call Waiting. If Call Waiting is allowed, Camp-On is not attempted. If Call Waiting is not allowed, Station-to-Station Camp-On is automatically attempted. If this succeeds, Priority Override can still follow. If Camp-On fails because there is no external call, Forced Camp-On and Priority Override may still work. However, if Camp-On fails because of other limitations, Forced Camp-On and Priority Override will also not work.

Even though Camp-On will still function when Warning Tone Denied (WTD) Class of Service is defined, Forced Camp-On and Priority Override require Warning Tone Allowed (WTA) Class of Service.

Priority Override is not allowed on analog (500/2500 type) telephones unless the Override Allowed (OVDA) Class of Service is defined. This Class of Service is also used for Override. This Class of Service does not affect Camp-On.

Camp-On requires an external call on hold. Forced Camp-On can be done without a call on hold, or with both internal or external calls on hold.

Trunks cannot perform Priority Override. They also cannot be overridden unless they are the unwanted party of a connection. It is for this exception that trunks are given a Priority Level.

New Camp-On Classes of Service (Camp-On From Another Telephone Allowed [CPFA], Camp-On From Another Telephone Denied [CPFD], Camp-On To Another Telephone Allowed [CPTA], and Camp-On To Another Telephone Denied [CPTD]) apply to Forced Camp-On and Automatic Forced Camp-On (AFCO) only. They do not apply to Station or Attendant Camp-On.

If a telephone is denied Forced Camp-On by Class of Service, Priority Override may still be attempted.

Feature interactions

Attendant Break-In

Telephones with a toll operator break-in call cannot be camped on to or overridden. Overflow tone is returned to telephones attempting either Forced Camp-on or Priority Override.

Attendant calls
Automatic Call Distribution

Telephones involved in Automatic Call Distribution (ACD) calls cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Call Hold, Deluxe
Call Hold, Permanent

Neither held calls nor telephones with calls on hold may be camped on or overridden. Overflow (fast busy) tone is returned to telephones attempting either a Forced Camp-On or Priority Override.

Camp-On

Station-to-Station Camp-On and Attendant Camp-On are not affected by Forced Camp-On or Priority Override. The following new Classes of Service affect only Forced Camp-On:

- Camp-On From Another Telephone Allowed (CPFA)
- Camp-On From Another Telephone Denied (CPFD)
- Camp-On To Another Telephone Allowed (CPTA)
- Camp-On To Another Telephone Denied (CPTD])

The Station Camp-On (SCMP) package 121 is required to return busy tone instead of ringback tone to the party camping on.

Conference calls

Telephones involved in conference calls cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

China – Attendant Monitor

A set may operate enhanced override on a desired call. If the desired call is being Attendant Monitored at the time, existing operation occurs for the first time the Enhanced Override key is pressed. The second time the key is pressed, the interaction with Attendant Monitor is the same as with regular override.

Data calls

Data calls have Warning Tone Denied (WTD) Class of Service, and cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting Forced Camp-On or Priority Override.

Digit Display

The Digit Display of the telephones being overridden changes to the Directory Number (DN) of the telephone overriding once Priority Override is accomplished.

Do Not Disturb

Telephones with Do Not Disturb (DND) enabled cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Hold

Neither held calls nor telephones with calls on hold may be camped on or overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Make Set Busy

Telephones with Make Set Busy active cannot be Forced Camp-On or Priority Override. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Multi-Party Operations

With Priority Override (POVR) equipped, there is a slight change in Multi-Party Operations functionality. When a consultation call is made without POVR equipped, and the telephone being called is busy, a recall returns to the party on hold without dialing a control digit. However, if POVR is equipped, a control digit must be dialed. Any control digit releases the busy call and returns to the call on hold.

Operator Call Back

Telephones involved in an Operator Call Back call or Toll Operator Break-In cannot be force camped on or priority overridden. Overflow (fast busy) tone is returned to telephones attempting either Forced Camp-On or Priority Override.

Override

If Priority Override is equipped, it replaces Override when using the OVR key or OVRD FFC. However, Override can be simulated by using the default PLEV, 2, for all trunk routes and telephones.

Ring Again

Ring Again (RGA) is the only other feature currently available once a busy telephone has been encountered. RGA is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Feature packaging

To provide the Enhanced Override capabilities, the following packages are required:

- Station Camp-On (SCMP) package 121
- Flexible Feature Codes (FFC) package 139
- Multi-Party Operations (MPO) package 141, and
- Priority Override/Forced Camp-On (POVR) package 186.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure the customer for Automatic Forced Camp-On.

2 LD 15 – Configure the customer for Station Camp-On tone.

3 LD 57 – Configure Override and Enhanced Override FFCs.

4 LD 10 – Configure analog (500/2500 type) telephones for Forced Camp-On, Priority, and Enhanced Override. Enter the Priority Override Level at the PLEV prompt.

5 LD 11 – Configure Meridian 1 proprietary telephones for Forced Camp-On, Priority, and Enhanced Override.Enter the Priority Override Level at the PLEV prompt. Define the override keys at the Key prompt.

6 LD 16 – Configure Route for Forced Camp-On, Priority, and Enhanced Override. Enter the Priority Override Level at the PLEV prompt

7 LD 14 – Allow a Warning Tone for trunks with Forced Camp-On, Priority, and Enhanced Override.

LD 15 – Configure the customer for Automatic Forced Camp-On.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	MPO	Multi-party options data block
...		
- AFCO	(NO) YES	Automatic Forced Camp-On. Enter YES if Forced Camp-On is to be applied automatically. Enter NO if Forced Camp-On is to be applied manually.

LD 15 – Configure the customer for Station Camp-On tone.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options data block

...		
- STCB	(NO) YES	Station Camp-On Busy tone. Enter NO if Busy Tone is not to be given to the transferring (controlling) party when the desired station is busy. Enter YES if Busy Tone is to be given to the transferring (controlling) party when the desired station is busy.

LD 57 – Configure Override and Enhanced Override FFCs.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	FFC	Flexible Feature Codes.
...		
CODE	x...x	Code to be programmed. Where x...x may be one of the following: OVRDOverride (OVRD is used for Priority Override when the Priority Override [POVR] package 186 is equipped.) EOVREnhanced Override (Is programmable only when the Priority Override [POVR] package 186 is equipped.)
X...X	y...y	The user is prompted with X...X, where X...X is the FFC code entered in response to the CODE prompt. y...y is a one-to-seven character input that the user must dial to use the FFC. Valid inputs are digits 0 through 9, asterisk (*), and octothorpe (#).

LD 10 – Configure analog (500/2500 type) telephones for Forced Camp-On, Priority, and Enhanced Override. Enter the Priority Override Level at the PLEV prompt.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
...		
CLS		Class of Service.
	(CPFA) CPFD	Forced Camp-On from another telephone to this telephone (allowed) denied.
	(CPTA) CPTD	Forced Camp-On to another telephone from this telephone (allowed) denied.
	OVDA	Override allowed.
	WTA	Warning Tone allowed.
...		
PLEV	0-(2)-7	Priority Override Level 0 Indicates that this telephone cannot be overridden or override. 1 Indicates that this telephone can be overridden but cannot override. 2 Indicates that this telephone can be overridden by telephones assigned level 2 through level 7 and that the telephone can override level 1 and level 2. 3-6 Similar to level 2, indicates that this telephone can be overridden by telephones assigned an equal or higher level and that it can override lower and equal levels, except level 0. 7 Indicates that this telephone can be overridden by another level 7 telephone only and that it can override level 1 through level 7.

LD 11 – Configure Meridian 1 proprietary telephones for Forced Camp-On, Priority, and Enhanced Override.Enter the Priority Override Level at the PLEV prompt. Define the override keys at the Key prompt.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = Meridian 1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS		Class of Service.
	(CPFA) CPFD	Forced Camp-On from another telephone to this telephone (allowed) denied.
	(CPTA) CPTD	Forced Camp-On to another telephone from this telephone (allowed) denied.
	WTA	Warning Tone allowed.
...		
PLEV	0-(2)-7	Priority Override Level. 0 Indicates that this telephone cannot be overridden or override. 1 Indicates that this telephone can be overridden but cannot override. 2 Indicates that this telephone can be overridden by telephones assigned level 2 through level 7 and that the telephone can override level 1 and level 2. 3-6 Similar to level 2, indicates that this telephone can be overridden by telephones assigned an equal or higher level and that it can override lower and equal levels, except level 0. 7 Indicates that this telephone can be overridden by another level 7 telephone only and that it can override level 1 through level 7.

...		
KEY		Define keys.
	xx OVR	Override (If Priority Override [POVR] package [186] is equipped, the OVR key is used for Priority Override.)
	xx EOVR	Enhanced Override (Allowed to be programmed only if Priority Override [POVR] package [186] is equipped.)

LD 16 – Configure Route for Forced Camp-On, Priority, and Enhanced Override. Enter the Priority Override Level at the PLEV prompt

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block
...		
PLEV	0-(2)-7	Priority Override Level. 0 Cannot be overridden. 1-7 Can be overridden by a telephone with a Priority Level that is equal to or greater than the level assigned to this route. ***Note:*** Trunks cannot override, but the levels of all parties in a connection are examined to determine if the connection may be overridden.

LD 14 – Allow a Warning Tone for trunks with Forced Camp-On, Priority, and Enhanced Override.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
...		
CLS		Class of Service.
	WTA	Warning Tone Allowed.

Feature operation

Forced Camp-On and Priority Override can be used when making either a simple call or consultation call (that is, having a call on hold while calling another party) call. The following feature operation descriptions use telephone A (an analog (500/2500 type) telephone) or telephone E (a Meridian 1 proprietary telephone) to call telephone B, which is connected to party C. Party D is used as the party on hold when either A or E is making a consultation call.

The telephones are configured as follows:

1 Telephone A is an analog (500/2500 type) telephone with Warning Tone Allowed (WTA) and Override Allowed (OVDA) Classes of Service.

2 Telephone B has Warning Tone Allowed (WTA) Class of Service.

3 Party C has Warning Tone Allowed (WTA) Class of Service and can be any telephone type or a Direct Inward Dial (DID), TIE, or Central Office (Public Exchange) (COT) trunk.

4 Party D can be any telephone or trunk.

5 Telephone E is a Meridian 1 proprietary telephone with Warning Tone Allowed (WTA) Class of Service and both an Override (OVR) and Enhanced Override (EOVR) key equipped.

For examples 1 to 4, assume the following:

1 Telephones A and E have a Priority Override Level (PLEV) of greater than “1”.

2 Telephones A and E both have Camp-On From Another Telephone Allowed (CPFA) Class of Service.

3 Telephone B and party C both have PLEVs greater than “0”, but less than or equal to those of telephones A and E.

4 Both telephone B and party C are involved in a simple call, not a conference call.

5 Telephone B has Camp-On To Another Telephone Allowed (CPTA) Class of Service.

6 Call Forward, Hunting, and Call Waiting are not in use.

Examples 1 to 4 are done with various combinations of Forced Camp-On and Priority Override. Forced Camp-On may be denied by responding "NO" to the Automatic Forced Camp-On (AFCO) prompt in LD 15, by configuring telephone E with only an Override (OVR) key and defining only the Override (OVRD) FFC in LD 57, or by setting the Classes of Service for both telephone A and E to Camp-On To Another Telephone Denied (CPTD) and Camp-On From Another Telephone Denied (CPFD). Both of these methods of disabling the Forced Camp-On feature do not affect the Priority Override feature. However, any conditions that would prevent Forced Camp-On from occurring also prevent Priority Override.

In the following feature operation descriptions, the term "recall" refers to a register recall, which may be performed in a number of different ways. Some typical examples are:

- Flashing the switchhook. This is the equivalent of hanging up the handset and picking it back up. This on hook, off hook action is performed in a time less than what the system would consider to be a valid disconnect.
- Pressing the flash or LINK button if equipped.

The Camp-On tone is always provided for Forced Camp-On, since Warning Tone Allowed (WTA) Class of Service is a prerequisite. This tone can be a buzz for Meridian 1 proprietary telephones or a single burst of tone for analog (500/2500 type) telephones if the customer option Periodic Camp-On Tone Denied (CTD) is selected in LD 15. If the customer option Periodic Camp-On Tone Allowed (CTA) is selected in LD 15, the Camp-On Tone as defined in the Flexible Tones and Cadences (FTC) LD 56 in response to the CAMP prompt will be used. The Priority Override tone used is the same tone as used for Override; this tone is defined in response to the OVRD prompt in the FTC LD 56.

While camping on, the party attempting the Camp-On, either telephone A or E, receives ringback if the Station Camp-On (SCMP) package 121 is not equipped, or receives either ringback or busy tone, as defined by the response to the Station Camp-On Busy tone (STCB) prompt in LD 15 if the SCMP package is equipped.

Override will take place on any established call when the Flexible Feature Code (FFC) is dialed or the Override (OVR) key is pressed. That means if telephone A calls telephone B while telephone B is busy and telephone B disconnects from that call and is established on another call when telephone A activates Override, the new call will be overridden.

Example 1
Enhanced Override with an analog (500/2500 type) telephone

With automatic Forced Camp-On turned off; Response to AFCO in LD 15 was "NO"

Table 6
Example of Enhanced Override with an analog telephone, with AFCO turned off.

STEP	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A receives busy tone.
3	A performs a recall.	A receives special dial tone (SDT).
4	A dials OVRD FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
	-or-	
4a	A dials EOVR FFC to attempt Forced Camp-On.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise B receives Camp-On tone and A receives ringback or busy tone depending on the options equipped. A is manually forced camped on to B.
4b	B disconnects from the call.	Telephone A rings telephone B.

	-or-	
4b	A performs a recall.	A receives SDT.
4c	A dials EOVR FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
5	If any party disconnects…	A simple two-party call is established.

With automatic Forced Camp-On turned on; response to AFCO in LD 15 was "YES".

Table 7
Example of Enhanced Override with an analog telephone, with AFCO turned on.

STEP	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A attempts Forced Camp-On to B.
2a	If Forced Camp-On was successful…	A receives ringback or busy tone depending on the options equipped. A is automatically forced camped on to B.
2b	B disconnects.	A rings B.
	-or-	
2a	A performs a recall and dials the OVRD or EOVR FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
	-or-	
2a	If Forced Camp-On was unsuccessful due to Class of Service restrictions…	A receives busy tone.

2b	A performs a recall and dials OVRD or EOVR FFC to attempt Priority Override. **-or-**	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.
2a	If Forced Camp-On was unsuccessful due to other limitations, then Priority Override is also restricted.	A receives busy tone.
2b	b) A performs a recall and dials OVRD or EOVR FFC to attempt Priority Override.	A receives overflow (fast busy) tone.

Example 2
Enhanced Override with a Meridian 1 proprietary telephone

With automatic Forced Camp-On turned off; response to AFCO in LD 15 was "NO".

Table 8
Example of Enhanced Override with a proprietary telephone with AFCO turned off.

STEP	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E receives busy tone.
3	E presses OVR key to attempt Priority Override. **-or-**	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.

3a	E presses EOVR key to attempt Forced Camp-On.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, B receives Camp-On tone and E receives ringback or busy tone depending on the options equipped. E is manually forced camped on to B.
3b	B disconnects from the call. **-or-**	Telephone E rings telephone B.
3b	E presses EOVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
4	If any party disconnects…	A simple two-party call is established.

With automatic Forced Camp-On turned on; response to AFCO in LD 15 was "YES".

Table 9
Example of Enhanced Override with a proprietary telephone with AFCA turned on.

STEP	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E attempts Forced Camp-On to B.
2a	If Forced Camp-On was successful…	E receives ringback or busy tone depending on the options equipped. E is automatically forced camped on to B.
2b	B disconnects. **–or–**	E rings B.

2a	E presses OVR or EOVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
	-or-	
2a	If Forced Camp-On was unsuccessful due to Class of Service restrictions…	E receives busy tone.
2b	E presses OVR or EOVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.
	-or-	
2a	If Forced Camp-On was unsuccessful due to other limitations, Priority Override is also restricted.	E receives busy tone.
2b	E presses OVR or EOVR key to attempt Priority Override.	A receives overflow (fast busy) tone.

Example 3
Enhanced Override from a consultation call with an analog (500/2500 type) telephone

With automatic Forced Camp-On turned off; Response to AFCO in LD 15 was "NO"; Station-to-Station Camp-On is denied or Station-to-Station Camp-On is equipped and D is a station; Multi-Party Operation is active.

Table 10
Example of Enhanced Override with an analog telephone with AFCO turned off

STEP	ACTION	RESPONSE
1	A is connected to D and B and C are connected in a simple call.	

2	A performs a recall.	A receives special dial tone (SDT). D is held.
3	A dials B.	A receives busy tone.
4	A releases.	Treated as misoperation of call transfer.
	-or-	
4a	A performs a recall and dials any control digit.	A releases from B and returns to D.
	-or-	
4a	A performs a recall.	A receives control dial tone.
4b	A dials OVRD FFC to attempt Priority Override.	Conference is established between A, B, and C with override tone given.
	-or-	
4a	A performs a recall.	A receives control dial tone.
4b	A dials EOVR FFC to attempt Forced Camp-On.	B receives Camp-On tone. A receives ringback or busy tone depending on the options equipped. A is manually forced camped on to B.
	– if –	
4c	A releases…	D is camped on to B.
	– if –	
4c	B disconnects.	A rings B.
	– if –	
4c	A performs a recall and dials any control digit.	A releases from B and returns to D.

	– if – A performs a recall and dials the POVR FFC again.	 If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.

With automatic Forced Camp-On turned off; response to AFCO in LD 15 was "YES"; Station-to-Station Camp-On is allowed and D is an external call; Multi-Party Operation active.

Table 11
Example of Enhanced Override with an analog telephone with AFCO turned off.

STEP	ACTION	RESPONSE
1	A is connected to D and B and C are connected in a simple call.	
2	A performs a recall.	A receives special dial tone (SDT). D is put on hold.
3	A dials B.	B receives Camp-On tone. A receives ringback or busy tone depending on the options equipped. A is automatically forced camped on to B.
4	A releases.	D is camped on to B.
	-or-	
	B disconnects	A rings B.
	-or-	
	A performs a recall and dials any control digit.	A releases from B and returns to D.
	-or-	

4a	A performs a recall.	A receives control dial tone.
4b	A dials OVRD or EOVR to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.

Example 4
Enhanced Override from a consultation call with a Meridian 1 proprietary telephone

With Automatic Forced Camp-On turned off; Response to AFCO in LD 15 was "NO"; Station-to-Station Camp-On is denied or Station-to-Station Camp-On is equipped and D is a station; Multi-Party Operation active.

Table 12
Example of Enhanced Override from a consultation call with a proprietary telephone. AFLO is turned off.

STEP	ACTION	RESPONSE
1	E is connected to D and B and C are connected in a simple call.	
2	E presses Conference or Transfer key.	E receives dial tone. D is put on hold.
3	E dials B.	E receives busy tone.
4	E releases or presses Conference or Transfer key again.	Treated as misoperation of call transfer.
	-or-	
	E presses the DN key that D is held on.	E is reestablished with D.
	-or-	
	E presses OVR key to attempt Priority Override.	Conference is established between E, B, and C with Override tone given.
	-or-	

4a	E presses EOVR key. **– if –**	B receives Camp-On tone. E receives ringback or busy tone depending on the options equipped. E is manually forced camped on to B.
4b	E presses Transfer key. **– if –**	D is camped on to B.
4b	B disconnects. **– if –**	E rings B.
4b	E releases. **– if –**	Camp-On is cancelled and E must press DN key to reconnect to D.
4b	E presses Conference or Hold key. **– if –**	Key operation is ignored.
4b	E presses the DN key that D is held on. **– if –**	E is reestablished with D.
4b	E presses EOVR key again.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.

With Automatic Forced Camp-On turned off; response to AFCO in LD 15 was "YES"; Station-to-Station Camp-On is allowed and D is an external call; Multi-Party Operation active.

Table 13
Example of Enhanced Override with proprietary telephone with AFCO turned off.

STEP	ACTION	RESPONSE
1	E is connected to D and B and C are connected in a simple call.	
2	E presses Conference or Transfer key.	E receives dial tone. D on hold.
3	E dials B.	E receives ringback or busy tone depending on the options equipped. E is automatically Forced Camped on to B.
4	E presses Transfer key.	D is camped on to B.
	-or-	
	B disconnects.	E rings B.
	-or-	
	E releases.	Camp-On is canceled and E must press DN key to reconnect to D.
	-or-	
	E presses Conference or Hold key.	Key operation is ignored.
	-or-	
	E presses the DN key that D is held on.	E is reestablished with D.

	-or- E presses EOVR or OVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.

Operation with various combinations of Forced Camp-On and Priority Override

The following tables show what happens when either Forced Camp-On or Priority Override are denied.

Forced Camp-On is denied by the new Camp-On From Another Telephone Denied (CPFD) and Camp-On To Another Telephone Denied (CPTD) Classes of Service.

Priority Override is denied for analog (500/2500 type) telephones by setting the Override Denied (OVRD) Class of Service, or for all telephones by setting their Priority Override Levels (PLEV) to 0.

Both Forced Camp-On and Priority Override are denied by the Warning Tone Denied (WTD) Class of Service, or if any of the limitations described in the Operating parameters or Feature interactions section is encountered.

The following table highlights the various combinations and the results of different actions for a simple call.

Table 14
Example of the results of various combinations of simple calls.

Setup						
AFCO setting in LD 15	NO	NO	NO	YES	YES	YES
Forced Camp-On Allowed	NO	NO	YES	NO	NO	YES
Priority Override Allowed	YES	NO	NO	YES	NO	NO
Action	**Result**					
A dials B B is busy	BT	BT	BT	BT	BT	BT or R
A recalls analog (500/2500 type) telephones only	SDT	SDT	SDT	SDT	SDT	SDT
A uses OVR key or OVRD FFC	POVR	O&L	O&L	POVR	O&L	BT or R
OR A uses EOVR key or FFC	BT	BT	BT or R	POVR	BT	BT or R
A uses EOVR key or FFC again	POVR	O&L	BT or R	POVR	O&L	BT or R

Legend:

BT – Busy tone returned to A.
BT or R – Busy tone or ringback returned to A; A camped on to B.
O&L – Overflow (fast busy) returned to A for 30 seconds, then A is locked out.
POVR – Priority Override is attempted.
SDT – Special dial tone is returned to A.

The following table highlights the various combinations and the results of different actions for a consultation call.

Table 15
Example of the results of various combinations of consultation calls.

Setup						
AFCO setting in LD 15	NO	NO	NO	YES	YES	YES
Forced Camp-On Allowed	NO	NO	YES	NO	NO	YES
Priority Override Allowed	YES	NO	NO	YES	NO	NO
Action	**Result**					
A connected to D A recalls analog (500/2500 type) telephones only	SDT	SDT	SDT	SDT	SDT	SDT
A dials B D is held. B is busy.	BT	BT	BT	BT	BT	BT or R
A recalls analog (500/2500 type) telephones only	CDT	CDT	CDT	CDT	CDT	CDT
A uses OVR key or OVRD FFC	POVR	O&R	O&R	POVR	O&R	BT or R
OR A uses EOVR key or FFC A uses EOVR key or FFC again	BT POVR	BT O&R	BT or R BT or R	POVR POVR	BT O&R	BT or R BT or R
A recalls analog (500/2500 type) telephones only	CDT	REC	CDT	CDT	REC	CDT
OR A presses DN key on which D is held	REC	REC	REC	REC	REC	REC

Legend:

BT – Busy tone returned to A.
BT or R – Busy tone or ringback returned to A; A camped on to B.
CDT – Control dial tone returned to A.
O&R – Overflow (fast busy) returned to A for 30 seconds, then A is reconnected to D.
POVR – Priority Override is attempted.
SDT – Special dial tone is returned to A; D is held.

If at any time invalid digits are dialed for the EOVR or OVRD FFC, overflow (fast busy) tone is returned to the telephone attempting to override. This telephone receives overflow (fast busy) tone for 30 seconds and is then locked out or reconnected to the telephone on hold. If the attempted override is made from a consultation call, the telephone may perform a recall during overflow (fast busy) tone, and return to the call being held.

Enhanced Override from a conference call with any telephone

Once a consultation conference (that is, party D is still on hold) has been established between telephone A or E and parties B and C, any of the following may occur.

Table 16
Example of Enhanced Override from a conference call.

ACTION	RESPONSE
Telephone B or C disconnects.	Telephone A or E remains in simple two party consultation with remaining telephone (B or C).
– or –	
Telephone A performs a recall and dials a control digit.	Multi-Party operation for control digit is dialed.
– or –	
Telephones B and C disconnect.	Telephone A or E may automatically be returned to telephone D or may have to perform a recall, depending on Class of Service (AO6/C6A and XFA). Override tone is removed.

– or – Telephone A disconnects or telephone E presses Transfer or Conference key.	D is transferred into the conference with B and C. Override tone is removed.
– or – Telephone E disconnects.	Telephones B and D remain connected. Telephone D is treated as in the case of misoperation of call transfer. Override tone is removed.

Override, Priority

Contents

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Feature description

The Priority Override feature allows users to break in to an established connection. To do this, analog (500/2500 type) telephone users enter the Override Flexible Feature Code (OVRD FFC), and proprietary telephone users use the Override (OVR) key.

Priority Override can be used as a feature by itself or in conjunction with Forced Camp-On. The combination of the two features is referred to as Enhanced Override (EOVR).

For Priority Override the overriding telephone must have a Priority Override Level (PLEV) that is greater than or equal to the PLEV of the telephone or trunk to be overridden.

For an analog (500/2500 type) telephone, a recall followed by dialing of the Priority Override FFC (OVRD FFC with Priority Override package 186 equipped) breaks into the connection and establishes a conference between all three parties. For a proprietary telephone, the OVR key is used in place of the FFC.

For Priority Override to be allowed, all telephones and trunks involved must have Warning Tone Allowed (WTA) Class of Service. Each telephone and trunk route (TIE, DID, and COT) is assigned a PLEV value:

Table 17
Priority Override Level Indication Assignments

PLEV	Indication
0	This telephone or route cannot be overridden. If assigned to a telephone, the telephone cannot use Override.
1	This telephone or route can be overridden. If assigned to a telephone, the telephone cannot use Override.
2	This telephone or route can be overridden by telephones assigned level 2 through level 7. If assigned to a telephone, the telephone can override level 1 and level 2.
3-6	(Similar to level 2) This telephone or route can be overridden by telephones assigned an equal or higher level. If assigned to a telephone, the telephone can override telephones assigned an equal or lower level, except level 0.
7	This telephone or route can be overridden by another level 7 telephone only. If assigned to a telephone, the telephone can override level 1 through level 7.

Operating parameters

Flexible Feature Codes (FFC) package 139 must be equipped for Priority Override to be available to analog (500/2500 type) telephones.

For analog (500/2500 type) telephone activation, Multi-Party Operations (MPO) package 141 must be equipped. Responsed with "YES" to the RALL prompt in LD 15 to ensure register recalls are required before dialing control digits. The OVRD FFC defined must not start with the same digit as one of the control digits. The control digits are defined in Overlay and are printed as part of the Customer Data Block (LD 21).

If Priority Override is equipped, it replaces Override when the OVR key or OVRD FFC is used. However, Override can be simulated by using the default value, 2, for all trunk routes and telephones.

Telephones or trunks involved in any of the following cannot be overridden:

- Non-established call
- Conference call
- Attendant call
- Attendant call using Centralized Attendant Service (CAS), Primary Rate Access (PRA), or Integrated Services Digital Network (ISDN) trunk
- Make Set Busy
- Do Not Disturb
- Automatic Call Distribution (ACD) call
- Operator Call Back
- Hold
- Data call
- Release Link call, and
- Parked call.

Priority Override is not allowed on analog (500/2500 type) telephones unless the Override Allowed (OVDA) Class of Service is defined. This Class of Service is also used for Override.

Trunks cannot perform Priority Override. They also cannot be overridden unless they are the unwanted party of a connection. It is for this exception that trunks are given a Priority Level.

Feature interactions

Attendant calls

Telephones involved in attendant calls cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Automatic Call Distribution (ACD)

Telephones involved in ACD calls cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Conference calls

Telephones involved in Conference calls cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Data calls

Data calls have Warning Tone Denied (WTD) Class of Service, and therefore cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Digit display

The Digit Display of the telephones being overridden changes to the Directory Number (DN) of the telephone overriding once Priority Override is accomplished.

Do Not Disturb (DND)

Telephones with DND enabled cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Hold

Neither held calls, nor telephones with calls on hold can be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Make Set Busy (MSB)

Telephones with MSB active cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Multi-Party Operations (MPO)

With Priority Override (POVR) equipped, there is a slight change in Multi-Party Operations functionality. When a consultation call is made without POVR equipped, and the telephone being called is busy, a recall returns to the party on hold without dialing a control digit. However, if POVR is equipped, a control digit must be dialed. Any control digit releases the busy call and returns to the call on hold.

Operator Call Back

Telephones involved in an Operator Call Back call or Toll Operator Break in cannot be Priority Overridden. Overflow (fast busy) tone is returned to telephones attempting Priority Override.

Override

If Priority Override is equipped, it replaces Override when using the OVR key or OVRD FFC. However, Override can be simulated by using the default PLEV, 2, for all trunk routes and telephones.

Ring Again

Ring Again (RGA) is the only other feature currently available once a busy telephone has been encountered. RGA is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Feature packaging

The Priority Override (POVR) feature is packaged under package 186. To provide all the capabilities described in this document, Flexible Feature Codes (FFC) package 139 should also be equipped.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 57 – Configure Priority Override FFC at the CODE prompt.

2 LD 10 – Configure Analog (500/2500 type) telephones for Priority Override.

3 LD 11 – Enter Priority Override levels and define override keys

4 LD 16 – Configure Route for Priority Override.

5 LD 14 – Configure trunks for Priority Override warning tones.

LD 57 – Configure Priority Override FFC at the CODE prompt.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	FFC	Flexible Feature Codes.
...		
CODE	OVRD	Change Override access code. OVRD is used for Priority Override when the Priority Override POVR package 186 is equipped.
OVRD	xxxx	Override access code.

LD 10 – Configure Analog (500/2500 type) telephones for Priority Override.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:		Type of telephone.
	500	analog (500/2500 type) telephone.
...		
CLS		Class of Service.
	OVDA	Override Allowed.
	WTA	Warning Tone Allowed.
...		
PLEV	0-(2)-7	Priority Override Level. 0 Indicates that this telephone cannot be overridden or override. 1 Indicates that this telephone can be overridden but cannot override. 2 Indicates that this telephone can be overridden by telephones assigned level 2 through level 7 and that the telephone can override level 1 and level 2. 3-6 Similar to level 2, indicates that this telephone can be overridden by telephones assigned an equal or higher level and that it can override lesser than and equal to levels, except level 0. 7 Indicates that this telephone can be overridden by another level 7 telephone only and that it can override level 1 through level 7.

LD 11 – Enter Priority Override levels and define override keys

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	xxxx	Telephone type, where: xxxx = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS		Class of Service.
	WTA	Warning Tone Allowed.
...		
PLEV	0-(2)-7	Priority Override Level. 0 Indicates that this telephone cannot be overridden or override. 1 Indicates that this telephone can be overridden but cannot override. 2 Indicates that this telephone can be overridden by telephones assigned level 2 through level 7 and that the telephone can override level 1 and level 2. 3-6 Similar to level 2, indicates that this telephone can be overridden by telephones assigned an equal or higher level and that it can override lesser than and equal to levels, except level 0. 7 Indicates that this telephone can be overridden by another level 7 telephone only and that it can override level 1 through level 7.
...		
KEY		Define keys.
	xx OVR	Override (If Priority Override [POVR] package 186 is equipped, the OVR key is used for Priority Override.)

LD 16 – Configure Route for Priority Override.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
PLEV	0-(2)-7	Priority Override Level 0 Cannot be overridden. 1-7 Can be overridden by a telephone with a Priority Level which is equal to or greater than the level assigned to this route. ***Note:*** Trunks cannot override, but the levels of all parties in a connection are examined to determine if the connection may be overridden.

LD 14 – Configure trunks for Priority Override warning tones.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
...		
CLS		Class of Service.
	WTA	Warning Tone Allowed.

Feature operation

Priority Override can be used when making either a simple or consultation call (that is, have a call on hold while calling another party). The following feature operation descriptions use telephone A (an analog (500/2500 type) telephone) or telephone E (a proprietary telephone) to call telephone B, which is connected to party C.

The telephones are configured as follows:

- Telephone A is an analog (500/2500 type) telephone with Warning Tone Allowed (WTA) and Override Allowed (OVDA) Classes of Service.
- Telephone B has Warning Tone Allowed (WTA) Class of Service.
- Party C has Warning Tone Allowed (WTA) Class of Service and can be any telephone type or a Direct Inward Dial (DID), TIE, or Central Office (Public Exchange) (COT) trunk.
- Telephone E is a proprietary telephone with Warning Tone Allowed (WTA) Class of Service and an Override (OVR) key equipped.

For the following descriptions:

- Telephones A and E have a Priority Override Level (PLEV) of greater than 1.
- Telephone B and party C both have PLEVs greater than 0, but less than or equal to those of telephones A and E.
- Both telephone B and party C are involved in a simple call, not a conference call.
- Call Forward, Hunting, and Call Waiting are not in use.

In the following feature operation descriptions the term “recall” refers to performing a register recall, which can be performed in a number of different ways. Some typical examples are:

- Flash the switchhook (the equivalent of hanging up the handset and picking it back up, this on hook, off hook is performed in a time period that is less than what the system would consider to be a valid disconnect).
- Press the flash or LINK button if equipped.

The Override tone is always provided for Priority Override since Warning Tone Allowed (WTA) Class of Service is a prerequisite. The Override tone used is the same tone as used for Override. The tone is defined in response to the OVRD prompt in LD 56.

Override will take place on any established call when the Flexible Feature Code (FFC) is dialed or the Override (OVR) key is depressed. That means if telephone A calls telephone B while telephone B is busy, and telephone B disconnects from that call and is established on another call when telephone A activates Override, the new call will be overridden.

Table 18
POVR with an Analog (500/2500 type) telephone

STEP	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	A dials B.	A receives busy tone.
3	A performs a recall.	A receives special dial tone (SDT).
4	A dials OVRD FFC to attempt Priority Override.	If telephone B or C has disconnected, telephone A receives overflow (fast busy) tone. Otherwise, a conference is established between A, B, and C with Override tone given.

Table 19
POVR with a proprietary telephone

STEP	ACTION	RESPONSE
1	B and C are connected in a simple call.	
2	E dials B.	E receives busy tone.
3	E presses OVR key to attempt Priority Override.	If telephone B or C has disconnected, telephone E receives overflow (fast busy) tone. Otherwise, a conference is established between E, B, and C with Override tone given.

Paging

Contents

This section contains information on the following topics:

Feature description 277
Operating parameters 278
Feature interactions 279
Feature packaging 279
Feature implementation 280
Feature operation 283

Feature description

The system provides switching access and trunk circuit interface to a customer-supplied speaker or radio paging equipment. Paging equipment is accessed by dial access or a Page key on Attendant Consoles. Telephones cannot be assigned a Page key and must dial access this feature.

Attendant Consoles using the Page key preempt telephones having only dial access. Telephones preempted by the attendant are disconnected and must re-access the paging trunk.

Time Forced Disconnect (TFD), provides a variable timer to force disconnect Paging trunks. The timer is defined on a route basis to limit the time a user can keep a Paging trunk seized. When the timer expires, the call is disconnected from the trunk. The trunk is disconnected when the Time Forced Disconnect (TFD) timer expires in all cases, regardless of the status of the trunk at the time. Timing starts as soon as the trunk is seized (not when the call is established), so the timer must allow some delay for connection time.

The Time Forced Disconnect timer is used on the following trunk types:

- COT Central Office
- DIC Dictation
- FEX Foreign Exchange
- PAG Paging trunks
- TIE Tie direct lines
- WAT Wide Area Telephone Service

Operating parameters

Station dial access to the Paging trunk is restricted by the Trunk Group Access Restriction (TGAR) code entered in LD 10 or LD 11.

Unique access codes are required for each Paging route.

Unique feature keys are assigned for each Paging route.

All Zone Paging is not available with the system, unless the customer-provided paging equipment is equipped with separate “all-zone” input.

The following requirements apply to Time Forced Disconnect (TFD) feature:

- The timer can only be assigned on a route basis and not to individual trunks. All trunks in a route have the same timer value.
- After a timer value is changed, it does not take effect on a given trunk until that trunk is released and seized again.
- Changing a timer value to zero (0) effectively removes the TFD timer from all the trunks in that route.
- The range of the timer is one hour, in 30-second increments (0–3600). The TFD timer is independent of all other timers.

Trunks forced off by TFD are disconnected normally, accompanied by an error message (ERR4054) output on the system terminal. The error message identifies the Originating Terminal Number (TN), Terminating Terminal Number (TN), date, and time for the following trunk types:

- Analog trunks
- Digital Trunk Interface (DTI) trunks, and
- ISDN Integrated Service Links (ISL)/Primary Rate Interface (PRI) trunks.

Feature interactions

Call Forward All Calls

Calls that originate on a TIE trunk to a telephone that is redirected to a paging route are blocked.

Conference

Paging trunks cannot be used in a conference call.

Multi-Party Operations

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a paging trunk.

Private Line Routes

Route 31 can be assigned as a paging route.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section.

1 LD 16 – Add or change a Paging trunk route access code and restriction group numbers.

2 LD 16 – Define the timer for the Time Forced Disconnect feature.

3 LD 14 – Add or change a Paging trunk within the Paging trunk route.

4 LD 12 – Assign Paging key for an Attendant Console. No programming is required to allow the attendant dial access to Paging.

5 LD 10 – Allow or deny dial access to Paging for analog (500/2500 type) telephones.

6 LD 11 – Allow or deny dial access to Paging for Meridian 1 proprietary telephones.

LD 16 – Add or change a Paging trunk route access code and restriction group numbers.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	PAG	Paging trunk route.
ICOG	OGT	Outgoing trunk.
ACOD	xxx...x	Trunk route access code (if the Directory Number Expansion package is equipped, this access code can have up to seven digits).

TARG	1-31	Trunk access restriction group number.

LD 16 – Define the timer for the Time Forced Disconnect feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
CNTL	(NO) YES	Changes to controls or timers (default is NO).
TIMR	TFD xxxx	TFD timer, where: xxxx = 0-(30)-3600 seconds, in 30-second increments.

LD 14 – Add or change a Paging trunk within the Paging trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PAG	Paging trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XUT XEM	Universal Trunk Card (NT8D14), E&M Trunk Card (NT8D15). Prompted only for superloops and the first unit on the card.
CUST	xx	Customer number, as defined in LD 15

SIGL	DX2 DX4 EAM EM4 LDR OAD	DX signaling (two-wire) – QPC71 only. DX signaling (four-wire) – QPC71 and NT8D15. E&M signaling (two-wire) – QPC71 and NT8D15. E&M signaling (four-wire) – QPC71 and NT8D15. Loop dial repeating – QPC71 and NT8D14/15. Outgoing automatic, incoming dial – QPC71, NT8D14/15.
STRO	IMM WNK DDL	Immediate start outgoing. Wink start outgoing. Delay dial outgoing.
SUPN	(NO) YES	Answer and disconnect supervision required.

LD 12 – Assign Paging key for an Attendant Console. No programming is required to allow the attendant dial access to Paging.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx PAG yyy...y	Paging key, where: xx = key number (0-9 on M1250, 0-19 on M2250), and yy...y = access code of Paging trunk route.

LD 10 – Allow or deny dial access to Paging for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

TGAR	xx	Allow/deny access to Paging trunk.

LD 11 – Allow or deny dial access to Paging for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
TGAR	xx	Allow/deny access to Paging trunk.

Feature operation

No specific operating procedures are required to use this feature.

Partial Dial Timing

Contents

This section contains information on the following topics:

Feature description 285
Operating parameters 286
Feature interactions 286
Feature packaging 286
Feature implementation 287
Feature operation 287

Feature description

This feature allows a partial dial timer to be associated with a Direct Inward Dialing (DID) route. The End-of-dialing timer is used for partial dial timing. It is defined on a route basis and has a range from 128 to 32640 milliseconds, in increments of 128 milliseconds.

The partial dial timer is started each time that a digit is expected. If the timer expires before a complete DN is dialed, the call is given treatments as shown in Table 20 on page 286.

Note: The Partial Dial Timing feature can be used with the End of Selection and End of Selection Busy features.

Table 20
Treatment of calls upon expiration of dial timer

PRDL EOS	NO	YES	BSY
NO	N/A	Call ATTN	Overflow tone
YES	N/A	EOS signal Call ATTN	EOS signal Overflow tone
BSY	N/A	EOS, EOSB signals Overflow tone	EOS/EOSB signals Overflow tone

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Partial Dial Timing feature described above.

The Partial Dial Timing feature is not available on 1.5 Mbit digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

The Partial Dial feature is not supported by R2 Multifrequency Compelled Signaling.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Create or modify partial dial timing for trunk routes.

Prompt	Response	Description
...		
PRDL	(NO) YES BSY	No partial dial timing on DID route, Partial dial timing is equipped using EOD, or Partial Dial timing is equipped using EOD; BSY signal is sent on time out.

Feature operation

No specific operating procedures are required to use this feature.

PE/EPE blocking

Contents

This section contains information on the following topics:

Feature description 289
Operating parameters 290
Feature interactions 290
Feature packaging 291
Feature implementation 291
Feature operation 291

Feature description

This feature introduces a blocking facility that disables all the cards, loops, and terminal numbers (TNs) configured on the PE/EPE shelves. This feature is active immediately after the install/upgrade.

The feature includes the following:

- A warning message displays during install/upgrade, which the user must acknowledge to complete the process.
- A reporting facility in LD 81 lists the hardware components that are blocked.
- A post blocking message displays if the user attempts to modify a TN or create a new TN on PE/EPE shelves.

Operating parameters

The operating parameters are as follows:

- If a TN is in a call, the call is dropped after the terminal is upgraded.
- A loop that is disabled before upgrade remains disabled after upgrade, and cannot be enabled. Meridian Mail TNs configured on this loop cannot be enabled.
- This feature does not affect Meridian Mail TNs, DTI loops, and PRI loops.
- Loop numbers on PE/EPE shelves cannot be reused until the blocks are manually removed.
- Even background loop tests and audits cannot enable the disabled units and cards.
- The user can **OUT** the existing TNs configured on PE/EPE shelves.

Feature interactions

This feature does not affect Meridian Mail TNs, DTI loops, or PRI loops configured on PE/EPE shelves. Once the hardware is disabled, no further handling on PE/EPE shelves is possible.

Station administration

The FEAT prompt in LD 81 accepts a new value, **PEPE**, that prints all the TNs, loops, and cards configured on PE/EPE shelves. Moreover a new **SCH1469** error message displays when the user attempts to change an existing TN or create a new TN on PE/EPE shelves.

Maintenance

OTM retrieves and displays the status of loops, cards, or units configured on the system. After the feature implementation, loops, cards, or units configured on PE/EPE shelves show a status of **PE/EPE disabled**.

The affected components are:

- Network loops

- PE shelves
- PE cards
- PE units

The **SCH1469** error message displays when the user attempts to change an existing TN or create a new TN on PE/EPE shelves.

Feature packaging

No new software packages are introduced for this feature.

Feature implementation

This feature is implemented immediately after an installation or upgrade to Succession 3.0 Software.

Feature operation

There are no specific procedures required to operate this feature.

Periodic Camp-on Tone

Contents

This section contains information on the following topics:

Feature description 293
Operating parameters 293
Feature interactions 294
Feature packaging 294
Feature implementation 294
Feature operation 296

Feature description

This feature replaces the single buzz or burst of tone for Meridian 1 proprietary telephones, given to indicate a camped-on call, with periodic bursts of buzz or tone. The buzz or tone can be defined on a customer basis.

The Periodic Camp-On Tone applies to calls camped-on by an attendant in standalone and Integrated Services Digital Network (ISDN) environments, and camped-on from inquiry calls in standalone environments.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Break-In
Attendant Busy Verify
Override

The Periodic Camp-On Tone has precedence over Break-In, Busy Verify, and Override intrusion tones.

Semi-Automatic Camp-On

Periodic Camp-On Tone stops when the camped-on call is recalled to the attendant.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP) package 131.

Dependency:

- Flexible Tones and Cadences (FTC) package 125

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 56 – Define a new cadence in the Master Cadence Table (if required).

2 LD 56 – Assign a cadence, either new or existing, to the Camp-On tone.

A tone with a periodic cadence must be defined for the Camp-On feature. An existing periodic cadence may be chosen from the Master Cadence Table, or a new cadence may be defined specifically for the Camp-On tone.

LD 56 – Define a new cadence in the Master Cadence Table (if required).

Prompt	Response	Description
...		
TYPE	MCAD	Master Cadence data block.
WCAD	0-225	Cadence Number to be given the new definition. Cadence number 0 is reserved for continuous tone and is not changeable.
CDNC	xxxx xxxx ... xxxx	Cadence. On-off phases for Cadence (ten off-on cycles). Entries 1 through 15 are reserved for ringing cadences. When defining the cadences in MCAD, each phase is entered in 5 millisecond increments. The first number defines the length of the first on period. The second defines the length of the first off period. The third defines the length of the second on period, and so forth. The range of the first phase is 1-9999 increments. The range of the second phase is 0-9999 increments. The default is 0 0 0 0 0 0 0 0 0 0.

LD 56 – Assign a cadence, either new or existing, to the Camp-On tone.

Prompt	Response	Description
...		
TYPE	FTC	Flexible Tones and Cadence data block.
CDNC	xxxx ... xxxx	The cadence number of the existing cadence, or the cadence number given to the newly defined cadence.
...		
SCCT	(NO) YES	Software Controlled Cadences and Tones. Modification of the software controlled definitions allowed.
- CAMP		Camp-On tone.
- - TDSH	i bb c tt	Tone definition for systems equipped with Tone and Digit cards, where: i = internal (0), or external (1) source bb = burst cc = cadence, and tt = frequency. Prompts with the response i bb c tt define the internal/ external source, burst, cadence and frequency/level respectively. Enter the decimal equivalent (0-15) of the TDS Hex code. The first field is usually 0. If an external source is used, the entry is 1 and the fourth field is 0-7 for the specified channel.
- - XTON	0-255	XCT tone code.

Feature operation

No specific operating procedures are required to use this feature.

Periodic Clearing

Contents

This section contains information on the following topics:

Feature description 297
Operating parameters 297
Feature interactions 298
Feature packaging 298
Feature implementation 299
Feature operation 299

Feature description

The Periodic Clearing Signal (PCS) is used to disconnect calls that have been answered, but are now either ringing, held (consultation hold), parked (on hold without consultation), or camped-on (in the process of being transferred to a busy extension). These calls receive PCS pulses that will serve to disconnect the call if the caller has hung up. If the caller is still waiting, the line remains connected. The Periodic Clearing feature includes a Disconnect Timer (DCTI) that indicates the time period (in seconds) before a call is disconnected. The timer can be used to disconnect a call even if the periodic clearing is disabled.

Operating parameters

This feature applies only to 2 Mbit digital incoming Public Switched Telephone Network (PSTN) and Direct Inward Dialing (DID) calls.

Feature interactions

AC15 Recall: Timed Reminder Recall

When the Periodic Clearing feature is active the Disconnect timer will interfere with the AC15 recall timer. The Disconnect timer is activated on a TIE trunk or an incoming Direct Inward Dialing (DID) or Central Office (CO) trunk which is connected to the AC15 TIE trunk. If the Disconnect timer expires first the AC15 recall is cancelled and the trunk is disconnected. This is the case with a call which has been established with a TIE trunk or an incoming call on a DID or CO trunk that has been extended over an AC15 TIE trunk with the timed recall activated.

Generic XFCOT Software Support

Periodic Clearing is the sending of periodic signal from the system to a Central Office when an incoming call has been answered but is not in an established state (for instance, ringing, held, or parked). The connection is disconnected if the originator goes on-hook.

The Periodic Clearing condition is timed by the disconnect timer (DCTI) to prevent this situation from lasting for an extended time. When the DCTI timer expires the trunk is disconnected.

The Disconnect Timer can be used without having the feature Periodic Clearing configured particularly when the Central Office trunk has no disconnect supervision. It can be disabled by setting the DCTI to 0 in LD 16.

A loop start trunk can be marked as disconnect supervised. When it has a class of service providing disconnect supervision, in Periodic Clearing condition the trunk is disconnected when the calling station releases the call.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Enable Periodic Clearing Signal for trunk routes at the PECL prompt.

Prompt	Response	Description
...		
PECL	(NO) YES	(Do not send) send Periodic Clearing signal.

Feature operation

No specific operating procedures are required to use this feature.

Periodic Clearing Enhancement

Contents

This section contains information on the following topics:

Feature description 301
Operating parameters 301
Feature interactions 302
Feature packaging 302
Feature implementation 302
Feature operation 302

Feature description

This feature permits the system to send a Periodic Clearing Signal (PCS) and/or start the Disconnect Timer (DCTI) on a TIE or TIE AUTO line, when a call is answered but not established to a station, and there is more than one analog (500/2500 type) telephone involved in the call.

The system can perform the following:

- Receive a PCS on a TIE trunk, then retransmit it to another TIE trunk or incoming 2.0 Mbps digital or analog Public Exchange/Central Office (CO) or Direct Inward Dialing (DID) trunk, and
- Start the DCTI for incoming Central Office or DID trunks.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Called Party Disconnect Control
Toll Operator Break-in

The Called Party Disconnect Control and Toll Operator Break-in can exist on the same system and function on the same routes, but are not to be used in conjunction with Periodic Clearing.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

LD 16 – Create or modify the length of ringing time allowed for trunk routes.

Prompt	Response	Description
...		
PECL	(NO) YES	(Do not send) send Periodic Clearing signal.
DCTI	(0)-511	The time (in seconds) an extension is allowed to ring or be on hold before the trunk is disconnected. 0 specifies disconnection will not occur.

Feature operation

No specific operating procedures are required to use this feature.

Periodic Clearing on RAN, Meridian Mail, ACD, and Music

Contents

This section contains information on the following topics:

Feature description 303
Operating parameters 304
Feature interactions 304
Feature packaging 304
Feature implementation 304
Feature operation 305

Feature description

This feature allows the periodic clearing signal to be sent in situations where an incoming call has been answered and connected to Meridian Mail, Automatic Call Distribution (ACD) queue, music, or a recorded announcement (including when the call has been forwarded to a pager, connected to Recorded Announcement (RAN), and placed in the pager queue). The periodic clearing signal is sent on incoming calls over Public Exchange/Central Office, Direct Inward Dialing (DID), TIE, 2.0 Mbps Primary Rate Interface (PRI2) TIE, and Integrated Services Digital Network Signaling Link (ISL) TIE trunks.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Called Party Disconnect Control

This feature is not supported if used together with Toll Operator Break-In.

Centrex Switchhook flash

This feature is not supported if used together with Centrex Switchhook flash.

Integrated Services Digital Network (ISDN) Basic Rate Interface

This feature is not supported on ISDN Basic Rate Interface.

MFC and MFE signaling

This feature is not supported if used on MFC and MFE signaling trunks.

Toll Operator Break-In

This feature is not supported if used together with Toll Operator Break-In.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP) package 131; and Network Attendant Service (NAS) package 159.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Allow or deny Periodic Clearing on Meridian Mail for a customer.

2 LD 23 – Configure Periodic Clearing on Meridian Mail.

LD 15 – Allow or deny Periodic Clearing on Meridian Mail for a customer.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	FTR	Features and options.
...		
OPT	(PCMD) PCMA	Deny (the default) or allow Periodic Clearing on Meridian Mail.

LD 23 – Configure Periodic Clearing on Meridian Mail.

Prompt	Response	Description
...		
PCMM	(NO) YES	Deny (the default) or allow Periodic Clearing on Meridian Mail. Prompted only if OPT = PCMA in LD 15.

Feature operation

No specific operating procedures are required to use this feature.

Periodic Pulse Metering

Contents

This section contains information on the following topics:

Feature description 307
Operating parameters 308
Feature interactions 308
Feature packaging 312
Feature implementation 312
Feature operation 314

Feature description

The Periodic Pulse Metering (PPM) feature allows the user of each station within a system to keep an accurate record of Public Switched Telephone Network (PSTN) and Direct Outward Dialing (DOD) calls for billing or administration. The PPM feature:

- Detects rapid PPM (the system will be able to detect and count at least three pulses per second).
- Records the accumulated PPM count for each call on the Call Detail Reporting (CDR) if equipped.
- Calculates and records the total charge for each call based on the assigned unit and the total number of received pulses for the call.
- Allows the attendant to mark a specified call in order to read out the number of accumulated PPM counts against this call.

- Allows the customer to specify a particular schedule for printing the MR reports.
- Supports Call Detail recording (CDR) on multiple call transfer for outgoing PPM calls.

Operating parameters

A Periodic Pulse Meter can count to a maximum of 32,767 pulses. When this limit is exceeded, an indication of overflow is provided.

To access message registration data, an SL-1 telephone must have a digit display. Hence, MR data access from the M2009, M2112, and M2018 is not supported.

The reading and changing of Periodic Pulse Meters from the M3000 telephone is not supported.

PPM is not supported by the 1.5 Mbit Digital Trunk Interface (DTI).

Feature interactions

AC15 Recall: Transfer from Norstar

If party Z (on Norstar) calls party X (an outgoing trunk with PPM or Advice of Charge on the system) and transfers the call to party Y, the call is charged against the AC15 trunk route's meter until the transfer is completed. When party Z completes the transfer in ringing status, the charges still accumulate in the AC15 trunk route's meter. If the call is in established status, the charges accumulate against party Y, if party Y has a meter. Otherwise, charges accumulate against the customer meter.

Advice of Charge for EuroISDN

Advice of Charge has the following interactions with the Periodic Pulse Metering (PPM): recording of accumulated call charging information for each call on the CDR record, calculating the total charge for each call based on the assigned unit cost and the accumulated information received from the network, allowing the attendant to read the number of call charge units on a per call basis and allowing a set with a MRK key to access Message Registration information.

Attendant Administration

Attendant Administration does not support the PPM feature.

Call Detail Recording

If both the Call Detail Recording (CDR) and Meter Registration feature are equipped for a customer, the PPM pulse counts for metered calls over trunks for which the CDR feature is enabled are recorded on the CDR record along with the standard CDR information. If the charge option is allowed, the charge for the call is calculated and recorded on the CDR. If the charge option is disabled, zeros are printed in the charge field on the CDR. As a customer option, the CDR records can be printed onto a teletype terminal or tape unit.

Call Forward All Calls
Call Forward No Answer
Hunting

Metered calls transferred or extended from one station to another using the Call Forward All Calls, Call Forward No Answer, or Hunting feature are charged against the last station at which the call is answered as the controlling station releases. The last party to forward a call onto a metered PPM trunk is charged.

Call Park

When a metered call is parked from one station to another, the controlling station is charged until the call is answered.

Call Pickup

Metered calls transferred or extended from one station and answered at another station using the Call Pickup feature are charged against the station where the call is picked up as the controlling party disconnects.

Call Transfer

If the user of a station which is connected to a metered trunk transfers an internal call to another internal station while the dialed station is still ringing, the PPM pulse count is accumulated against the transferring station until the call is answered by the dialed party, or abandoned by the dialing party. When the call is answered, the pulses are counted against the station to which the call has been transferred.

If the station user transfers the call after consulting with the dialed station user, then the PPM pulses are counted against the controlling station until the call is transferred. When the call is transferred, the PPM pulses are counted against the station to which the call has been transferred. If the transferred call is redirected using any of the call redirection features such as Call Forward or Hunting, the call is charged against the transferring station until the call is transferred. The pulses are then counted against the answering station. This method ensures that PPM meters are charged in a manner consistent with the printing of CDR records.

Camp-On

Metered calls camped-on to a busy station by an attendant are charged against the attendant until the call is answered and the attendant releases.

Conference - Attendant

If an attendant establishes a conference which includes one or more metered trunks, and the attendant first dials a metered trunk as a source, the PPM pulses are counted and accumulated against the attendant. If the attendant continues to hold the conference at the console, the pulses continue to accumulate against the attendant. If the attendant releases the conference from the console, the pulses are accumulated against the station that has been in conference the longest. If the attendant first dials an internal station or a TIE trunk, any connection established thereafter is charged against this station or trunk.

Conference - Three-party/Six-party

Whenever a PPM trunk is added to a conference, a CDR Start record is generated, if CDR is equipped on the trunk. The PPM pulse counts from the trunk are accumulated against the party who initiated the call. If a party who adds a PPM trunk to the conference disconnects while the conference is still in progress, read requests are sent to the PPM trunk to read the residual count. Then, the on-board counter is cleared, the residual count is added to the temporary meter, and the contents of the temporary meter are added to the terminal meter. A CDR Transfer (X) record is then printed against this party, and the temporary meter is cleared. The party that is charged is the one that has been in conference the longest. When a trunk with disconnect supervision disconnects, a CDR End record is immediately printed. For trunks that do not

provide a disconnect signal, their CDR records are not printed until the last party disconnects from the conference.

Consultation calls

If a user establishes a consultation call including one or more metered trunks, all the associated pulses are counted against the controlling station until the call is transferred.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Periodic Pulse Metering is not supported by CIS DTI.

Italian Central Office Special Services

Periodic Pulse Metering pulses are received from the Central Office according to the charge of the accessed service, and are collected and stored as per normal procedures.

Italian Periodic Pulse Metering

This feature now allows PPM pulses to be counted on Italian DTI2 trunks. The Italian DTI2 option default is set to NA (that is, not active when software prior to the introduction of this feature is upgraded). Existing operation thus continues unaffected by the new feature.

Recall to Same Attendant

Meter recalls are returned to the same attendant whether Recall to Same Attendant is allowed or not. If Return to Same Attendant with Queuing on Busy (RSAQ) is selected as an option, the recalls are queued to a specified attendant.

Tandem Switching

If an incoming TIE trunk is connected to a PPM trunk, the pulses are counted against the access code of the TIE route.

Virtual Network Service

Periodic Pulse Metering is supported on the Virtual Network Service Bearer trunks only.

1.5 Mbps Digital Trunk Interface

PPM is not supported by 1.5 Mbps DTI.

2 Mbps Digital Trunk Interface

PPM operates the same for 2 Mbps DTI as for analog trunks.

Feature packaging

This feature is packed under Periodic Pulse Metering/Message Registration (MR), package 101.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Select PPM functionality in the Configuration Record.

2 LD 12 – Create or modify a meter key for Attendant Consoles.

3 LD 15 – Assign Meter Incoming Call Indicator.

4 LD 16 – Create or modify data for each DID trunk route data block to allow or deny MFC Signaling option.

5 LD 14 – Polarity Sensitivity of Trunk Data Blocks must be created or modified.

LD 17 – Select PPM functionality in the Configuration Record.

Prompt	Response	Description
REQ	CHG	Change
TYPE	PARM	System parameters
...		
MTRO	PPM	Periodic Pulse Metering meter option

LD 12 – Create or modify a meter key for Attendant Consoles.

Prompt	Response	Description
...		
KEY	xx MTR	Add a meter key.

LD 15 – Assign Meter Incoming Call Indicator.

Prompt	Response	Description
...		
ICI	xx MTR	xx is the selected key/lamp number.

LD 16 – Create or modify data for each DID trunk route data block to allow or deny MFC Signaling option.

Prompt	Response	Description
...		
CDR	(NO) YES	Call Detail Recording for the trunk route.
MR	PPM	Message Registration Buffered PPM signal to be counted on this route.

LD 14 – Polarity Sensitivity of Trunk Data Blocks must be created or modified.

Prompt	Response	Description
...		
SIGL	GRD LOP	Signaling start arrangement, Ground or Loop.
SUPN	YES (NO)	Trunk Supervision required (not required)
STYP	PSP (PIP)	Polarity sensitive packs. Polarity insensitive packs.

Feature operation

If the attendant desires billing information immediately upon the completion of a long distance call, the call must be flagged by the attendant as a metered call. When a metered call is terminated or modified, the same attendant is recalled and the calculated call charge or PPM count for this call is displayed on the console. If the call is transferred, a Meter Recall will be routed to the attendant for each portion of the trunk connection.

The following keys are added the Attendant Console for this feature:

- The **MTR** key and lamp that can be assigned at any position on the flexible feature key strip on the Attendant Console, and
- The **Meter Recall ICI** key and lamp that can be assigned at any ICI position on the Attendant Console.

Marking a Call as Metered

The attendant can request the call charge or PPM count on any outgoing PPM call by pressing the **MTR** key after the PPM call has been made. When the **MTR** key is pressed, the meter lamp is lit and all the metered outgoing PPM trunks connected to the active console loop (for example, as in a conference) are marked as metered. Additional PPM trunks added to the conference hereafter are marked as metered automatically. Metering a non-PPM call is ignored.

To cancel the metered flag press the **MTR** key.

Meter Recall

When a metered call is modified or disconnected, a meter recall is presented to the attendant. The following occurs:

1 The meter recall ICI lamp comes on.

2 The Source side of an idle loop is lit.

3 The Destination lamp remains off.

4 The following information appears on the display of the Attendant Consoles:

a. The DN of the station or Access Code of the TIE trunk on which the external call was placed is shown on the left-hand portion of the digit display.

b. If the option “charge to Attendant Console” is selected, the call charge is calculated by multiplying the PPM count in the temporary meter for this call by the customer assigned unit cost. The call charge is then shown on the right-hand portion of the digit display. If an overflow occurs when the charge is calculated, an overflow indication is given to the attendant – “DN-32767”.

c. If the option “charge to Attendant Console” is disabled, the PPM count in the temporary meter is shown on the right-hand portion of the digit display.

If the attendant who originated the metered call is in Position Busy, the meter recall is presented to the next idle Attendant Console. It is possible for an Attendant Console unequipped with a **MTR** key to receive a meter call. If all attendants are in Night Service or Position Busy, the recall is saved in the attendant queue until one of the attendants becomes idle.

An attendant answers the meter recall by pressing the **Loop** key, and releases the Call by pressing the **Rls** key or another **Loop** key.

Personal Call Assistant

Contents

This section contains information on the following topics:

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Feature description

Personal Call Assistant (PCA) is a virtual Terminal Number (TN) that is placed in a Multiple Address Directory Number group. PCA can extend and ring calls to any dialable destination, thus enabling the creation of MADN groups across networks. If two calls arrive at the same time, the first caller is processed as outlined above, and the second caller receives busy treatment according to the current functionality of MADN.

Note: An attendant console cannot be the target of PCA.

PCA enables the simultaneous ringing of clients with different Directory Numbers (DNs) that are dispersed across a network. When simultaneous ringing is enabled, the Calling Line Identification (CLID) of the originating set is presented to the called set, provided CLID is available to PCA.

To support simultaneous ringing, MADN functionality has been enhanced as follows:

- MADN groups can exist across networks; end users can be located at any dialable location and have a different Directory Number (DN) than the MADN.
- The system can extend MADNs to local numbers if the switch is configured to forward to external numbers.
- Signaling is extended to the target node from the system to enable users on the system to access multi-media applications.

Every call associated with a target node user consists of an incoming call and an outgoing call that are joined after the call is answered. CDR records, pegs, and logs reflect these call dynamics.

If the original call is abandoned, the desktop stops ringing and PCA releases the extended call.

If a call is answered coincidentally on the target node desktop, then call treatment is queue-dependant as the messages are processed in the order in which they are received. If the trunk call is processed last, the call is released and the trunk is dropped. If the desktop call is processed last, dialtone is presented to the desktop.

A call that is not answered continues to ring until it either receives redirection treatment or is abandoned by the originator. The redirection treatment can be node-system initiated or can be the called party's arrangement on a residential or cell phone.

Call presentation to the desktop can be impacted by the Busy and Redirection treatment specified in the target (for example, in the cell phone).

PCA functionality enables multi-media capabilities on Succession 1000M, Succession 1000, and Meridian 1 systems that were previously restricted to the Succession Multimedia Xchange (MX).

The PCA feature supports the following telephones and terminals:

- analog (500/2500-type) telephones

- digital telephones
- IP terminals and clients (i2002, i2004, i2050 [soft client])

PCA can extend the call across all trunk types; however, CLID is supported only on ISDN PRI, PRI2, and H.323 trunks.

Operating parameters

The Personal Call Assistant feature applies to the following systems with Succession 3.0 and later software:

- Succession 1000
- Succession 1000M Half Group
- Succession 1000M Chassis
- Succession 1000M Single Group
- Succession 1000M Multi Group
- Succession 1000M Cabinet
- Meridian 1 Option 61C CP PII
- Meridian 1 Option 81C CP PII
- Meridian 1 Option 11C Cabinet
- Meridian 1 Option 11C Chassis

When multiple PCAs are configured with the same MADN, you cannot update individual PCAs using the FFC.

Network wide FFC operation is not supported; however, FFC does work with DISA.

FFCs cannot be invoked from the attendant console.

The HOT P DN length can be only 21 characters due to the space restrictions in the Call Register.

> ***Note:*** If the HOT P DN is updated using LD 11, the DN length can be a maximum of 32 characters.

Feature interactions

The PCA feature has the same feature interactions as the MADN feature, which are as follows.

Automatic Redial

An ARDL call from a Single Call Ringing (SCR) or Single Call Non Ringing (SCN) is redialed only when all sets that have the same DN are free.
An ARDL call from a Multiple Call Ringing (MCR) or Multiple Call NonRinging (MCN) is redialed only when the originating key is free.

Automatic Wake Up

All Multiple Appearance DNs are rung, including both primary and secondary DNs. Programming the wake up request using the Wake Up key applies only to telephones with the primary DN on key 0, and the Wake Up indicator operates as described only on the telephone that is currently programming the wake up request.

In addition, if two or more Multiple Appearance Primary DN telephones program a wake up request at the same time, the last telephone to finish overrides. All telephones with the same primary DN get the same request time of the last telephone to program a request. If the last telephone cancels the request, all requests are canceled. When the wake up programming sequence is finished, all Wake Up indicators on Multiple Appearance Prime DNs are updated unless a telephone is in the middle of Wake Up programming. If the AWU Recall option is chosen, the recall is presented to any idle attendant console in the same Console Presentation Group (CPG) equipped with the AWU key.

Note: An attendant console cannot be the target of PCA.

Automatic Wake up FFC Delimiter

For Multiple Appearance Directory Numbers, wake up information is stored, deleted, and queried from a DN's first primary appearance Terminal Number.

Call Detail Recording

CDR for Personal Call Assistant is handled in the following manner:

- If the call is answered on the desktop, there is no change from existing CDR operation.
- If the call is answered on a cell phone supported by the Succession MX, the following CDR records are created:
  - Succession 1000M, Succession 1000, and Meridian 1: PCA to DOD (Succession MX) and DID to DOD (after the call is joined to the Succession MX)
  - Succession MX: DID (from Succession 1000M, Succession 1000, and Meridian 1) to DOD (cell phone)

Call Detail Recording on Redirected Incoming Calls

If the DN of the set forwarding the call is a Multiple Appearance DN, the Terminal Number of the set is printed out in the AUX ID field (that is, line two of the Call Detail Recording record).

Call Forward by Call Type
Call Forward No Answer, Second Level

Call redirection parameters such as Call Forward No Answer are derived from the TN data block of the prime appearance of the called MADN. If there is more than one prime appearance, the parameters are selected from the last TN in the DN block. If more than one prime appearance of the MADN exists, the following information must be considered prior to configuring call redirection parameters for MADNs.

The DN Block organizes MADN information in numerical TN order. The TN with the highest numerical value (000-0-06-03) is placed at the beginning of the list. The list then continues in descending order with the lowest numerical TN (000-0-03-01) at the end of the list. Service change activity affects the organization of the DN list as described in the following paragraphs.

- If a telephone undergoes Service Change, its TN is moved to the beginning of the DN list, irrespective of the numerical value. This telephone remains at the beginning of the list until another service change or a SYSLOAD.

- If a DN appears on analog (500/2500-type) telephones, and digital telephones, the analog (500/2500-type) telephones are listed in numerical TN order at the top of the list. Digital telephones are listed in numerical TN order at the bottom of the list. A Service Change to an analog (500/2500-type) telephone moves its TN to the beginning of the list. A Service Change to a digital telephone moves its TN to the end of the list.
- A SYSLOAD restructures the list back to numerical TN order, with analog (500/2500-type) telephones at the top and digital telephones at the bottom. Call redirection parameters continue to be derived as described in the preceding paragraphs.

Call Forward, Remote (Attendant and Network Wide)

The Call Forward, Remote (RCFW) feature applies only to the primary appearances of Multiple Appearance DNs, and it is recommended that only one appearance of a Multiple Appearance DN be configured as the prime DN. For the case of multiple stations with the same prime DN and SCPW, the RCFW operation applies to the station that has the Multiple Appearance Redirection Prime (MARP) assigned to it.

If none of the stations having the DN and SCPW assigned are configured as the MARP TN for that DN, the RCFA and RCFD applies to all stations matching the DN and SCPW. The attendant-based RCFW feature applies remote call forward operation only to the prime DN with MARP status. If the DN is not the prime DN or does not have MARP status, the user receives overflow tone.

Call Waiting Redirection

The Call Waiting Redirection feature applies to unanswered Call Waiting calls that apply to single appearance DNs and primary appearance DNs of MADNs.

Calling Party Name Display Denied

For a ringing call to a Multiple Appearance DN, the name on the calling set display can be suppressed by configuring any of the Terminal Numbers with NAMD Class of Service. The digit display on the calling set cannot be suppressed — the called digits are displayed even though the Class of Service on any of the Terminal Numbers is DIGD. The called set display is subject to the Class of Service of the calling party. For an established call to a Multiple Appearance DN, the calling set display is subject to the Class of Service configured for the answering set. The answering set display only is subject to the Class of Service of the calling party – the displays of the other sets in the Multiple-appearance group are blank.

China – Attendant Monitor

If Attendant Monitor is attempted on a Multiple Appearance DN, the Multiple Appearance Redirection Prime (MARP) TN becomes the desired party.

Controlled Class of Service

Controlled Class of Service (CCOS) restriction levels are activated or canceled on controlled telephones through their Prime Directory Number (PDN). When the PDN of a digital telephone is made CCOS active, all DNs on that telephone are also restricted. If the DN is a PDN on other telephones, those telephones are also restricted (if they have CCSA Class of Service).

Controlled Class of Service, Enhanced

All Controlled Class of Service (CCOS) restriction levels are activated and canceled from the Prime Directory Number (PDN) for CCOS controlling telephones. The PDN for an SL-1 telephone is made CCOS active, and all DNs for that telephone are restricted as well. If that DN is a PDN on other telephones, they are also restricted (if they have CCSA Class of Service).

Digital Private Signaling System #1 (DPNSS1) Executive Intrusion

If the attendant tries to extend a call to a DN that appears on more than one set, this DN can either be:

- Multiple-Call Arrangement with Ringing (MCR). When a call terminates on this DN, all idle stations on which the DN appears are rung. The call is established only with the station which has answered first. All others are idle.
- Multiple-Call Arrangement with No Ringing (MCN). The only difference between MCN and MCR is that the called stations are not rung (only their DN keys flash).
- Single-Call Arrangement with Ringing (SCR). When a call terminates on this DN, all idle stations on which the DN appears are rung. The call is established only with the station which has answered first. All others are busy.
- Single-Call Arrangement with No Ringing (SCN). The only difference between SCN and SCR is that the called stations are not rung (only their DN keys flash).

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Since the ANI category is defined on a per-set basis, two stations with the same Multiple Appearance Directory Number (MADN) can be assigned different ANI categories.

Directory Number Expansion

The DN can have up to seven digits if the Directory Number Expansion package is equipped. If Loop Restriction Removal is allowed, telephones with MADNs can be moved across loops using Automatic Set Relocation (LD 25), the digital telephones data block (LD 11), the analog (500/2500-type) telephone data block (LD 10), or Attendant Administration.

Display Calling Party Denied

When a Multiple Appearance DN is ringing, the display of the calling telephone does not show the caller's name if at least one of the TNs has Named Denied (NAMD) Class of Service. The dialed DN displays even if one TN has a DN Denied (DDGD) Class of Service. The display of the called telephone shows the DN and the caller's name according to the Class of Service of the calling DN. When a Multiple Appearance DN is answered, the display of the calling telephone shows the DN and caller's name and DN according to the Class of Service of the answering TN. The display of the answering telephone remains the same, while the displays of the other telephones are blanked.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The same locked or unlocked state applies to all Terminal Numbers with the same primary DN and the same SCPW. Terminal Numbers with the same DN, but not having the same SCPW, cannot be locked or unlocked.

Group Call

The maximum number of DNs that can be added as members of a Group Call is 20. Each Multiple Appearance, MCR/MCN DN reduces the number of telephone sets that can be added to a Group Call. For example, if two telephones have the same MCR appearance of a DN, the number of telephones in the Group Call becomes 19. That is, each appearance of a DN counts as one member, up to a maximum of 20, of the Group Call.

> ***Note:*** Multiple Appearance, SCR/SCN DNs count as one member of a Group Call, irrespective of its number of DN appearances.

Group Hunt

While Multiple Appearance DNs (MADN) single call arrangements are treated the same as Single Appearance DNs (SADN), MADN multiple call arrangements must be avoided in a group hunt list. With MADN multiple call arrangement, the idle or busy status of the MADN is determined by the Terminal Number (TN) data block of the prime appearance of the called DN. If there is more than one prime appearance of the called DN, the idle or busy status is then selected from the last TN in the DN block for the MADN (DNB prompt in LD 22). This means that there can be idle appearances of the MADN, while the hunt cycle regards them as busy and attempts to terminate on the next idle member of the group hunt list.

If an MADN multiple call arrangement must be used, a supervisor set must be assigned to the hunt group. This supervisor set must be given the only prime appearance of the MADN. Any other appearance must have the MADN programmed as a secondary DN (any DN key other than 0). In this way, the supervisor set controls the status of the MADN and thus the group hunt treatment. If the supervisor set is busy, the hunt does not terminate on the MADN.

Hunting

Hunting can be controlled by the MADN Redirection Prime (MARP) Terminal Number (TN). If the MARP system option is disabled, Hunting proceeds as if MARP did not exist. If all the telephones in the Multiple Appearance Directory Number (MADN) group are digital telephones, ringing telephones are placed at the top of the DN list, and non-ringing telephones are placed at the bottom.

If a Multiple Appearance Directory Number appears in a group with several telephone types, the telephone type affects the position of the TN in the list. The analog (500/250-type) telephones are listed at the top, and digital telephones are listed in numerical TN order at the bottom. A service change to an analog (500/2500-type) telephone moves its TN to the top of the list. A service change to a digital telephone moves it to the bottom of the list.

Call redirection follows the TN order from top to bottom. The MARP TN is always checked to determine if and how the call is to be redirected by Hunting, regardless of where the MARP TN resides in the TN list of the DN block. No searching of the TN list of the DN block is needed.

Hunting follows the hunt chain based on the originally dialed DN. The actual functioning and requirements for Hunting are not changed by the MARP feature. The basic change introduced by the MARP feature is to always have a designated TN, the MARP TN, as the TN supplying the call redirection parameters. If the MARP TN does not have Hunting control enabled, no Hunting is attempted. Other features for redirecting calls to busy DNs may be attempted based on the MARP TN.

A Short Hunting sequence begins when the MARP TN of a busy DN can perform Short Hunting. When a Short Hunt begins, it completes on that telephone before going to the Hunt DN. The precedence of Short Hunting over normal Hunting is maintained. Once a Short Hunting sequence is started on a digital TN, all the DNs in the Short Hunt sequence on that TN are attempted before redirecting the call to the TN's Hunt DN. Thus, a Hunt Chain connects Short Hunting sequences through Hunt DNs only.

Last Number Redial

A last number dialed on a Directory Number (DN) with multiple appearances is stored only against the telephone from which the number was originally dialed.

Loop Restriction

If Loop Restriction removal is not enabled, telephones with MADNs can be moved by using the Automatic Set Relocation feature (LD 25) or the Attendant Administration feature.

Meridian 911

The DN keys for multiple appearance sets can be defined as an SCR (single call ringing) key or as an MCR (multiple call ringing) key. For those DNs (keys on MADN sets) that are SCR, only one call can be answered at a time. That is, once a call taker answers a call, future calls to that DN receive busy tone until the call taker on that DN disconnects. For DNs that are MCR, calls are given busy tone once every call taker is busy answering a call. If one call taker is answering a call and there are other call takers available, a new call to that DN causes the sets of the available call takers to ring. Any available call taker can then answer the new call.

Message Registration

For Multiple Appearance Directory Number (MADN), the system selects the appropriate meter for the DN as follows. The MADN accesses the meter of the most recently configured telephone having a Prime DN (PDN) appearance and Message Registration Allowed (MRA) Class of Service. If no Terminal Number (TN) in the DN block has MRA Class of Service, the customer meter is charged. For the Message Registration Key (MRK), the system provides overflow and sets the MRK lamp to flash. For the Background Terminal (BGD), it prints a NO DATA FOUND message.

Privacy

If a Multiple Appearance, Single Call Arrangement (SCR) or Single Call Arrangement without Ringing (SCN) DN is shared by digital telephones only, Privacy is in effect. No one can enter a call unless the call is first placed on Hold, or unless Privacy Release is activated to enable another appearance to enter the call. If this configuration is shared between these telephones and single-line telephones, Privacy is not in effect for any appearance of the DN. Anyone sharing the DN can enter the call at any time.

Privacy Override

Since the Privacy feature is not active in this mode, telephones with a Privacy Override Denied Class of Service can bridge into an active call.

Privacy Release

Privacy Release has no effect on Multiple Appearance, Multiple Call Arrangement with Ringing (MCR), or Multiple Call Arrangement without Ringing (MCN) calls.

Remote Call Forward

With a Multiple Appearance Directory Number (MADN) and both sets having a Station Control Password (SCPW), Remote Call Forward does not operate as intended. That is, if Call Forward has been activated using the Remote Call Forward feature, Call Forward remains activated when an attempt to deactivate it is made from the set on which it is active.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Since the ANI category is defined on a per-set basis for Three Wire Analog Trunks, two stations with the same multiple Appearance DN can be assigned different ANI categories.

Voice Call

If a Voice Call DN is added to a second telephone, the DN becomes a Multiple Appearance DN (MADN). Voice Call does not support MADN. In addition to MADN interactions, PCA has the following feature interactions.

Feature packaging

This feature introduces the following package:

- Personal Call Assistant (PCA) package 398

Feature implementation

Task summary list

The following is a summary of tasks in this section:

- LD 15 – Enable PCA at the customer level.
- LD 57 – Configure FFCs for PCA control.
- LD 11 – Configure a new PCA.
- LD 97 – Add virtual superloops.

LD 15 – Enable PCA at the customer level. (Part 1 of 2)

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Features and options

LD 15 – Enable PCA at the customer level. (Part 2 of 2)

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15
...	...	...
VO_ALO	(NO) YES	Enable (disable) Virtual Office Automatic Logout.
PCA	(OFF) ON	Enable (disable) Personal Call Assistant Configuration of the PCA is preserved and enabled regardless of whether or not the feature is enabled.
TPDN	yyy.y	Target PCA DN Where: yyy.y is the primary DN TPDN is prompted only if PCA is set to ON. If there is no DN configured against the HOT P key in LD 11, this value is used to extend the call using the PCA feature. Enter X to remove. However, if there is at least one PCA with no target DN configured in LD 11, then this operation does not succeed.

LD 57 – Configure FFCs for PCA control. (Part 1 of 2)

Prompt	Response	Description
REQ	NEW CHG PRT	Create, change, or print a data record.
TYPE	FFC	Flexible Feature Code
CODE	PCAA	This is the code to activate PCA or change the HOT P DN.
PCAA	xxxx	Code number
CODE	PCAD	Code to deactivate PCA
PCAD	yyyy	Code number

LD 57 – Configure FFCs for PCA control. (Part 2 of 2)

Prompt	Response	Description
CODE	PCAV	Code to verify the status of PCA
PCAV	ZZZZ	Code number

LD 11 – Configure a new PCA.

Prompt	Response	Description
REQ:	NEW CHG	Add or change a PCA.
TYPE:	PCA	Personal Call Assistant
TN		Terminal Number
	l s c u	For Large Systems
	c u	For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
CLS	AHA	Automatic Hold Allowed (AHA). AHA is configured by default when the response to the TYPE prompt is PCA.
KEY	0 aaa yyy.y	Primary PCA DN Where aaa = MCN, MCR, SCN, or SCR and yyy.y = the primary DN Note: The PCA should never be configured as a MARP in a MADN group.
	1 HOT P x vvv.v	Target PCA DN where x = length of target DN vvv.v = the target DN Note: HOT P key is the default key. This key must be configured by the user.

LD 97 – Add virtual superloops.

Prompt	Response	Description
REQ	CHG	Add virtual superloops.
TYPE	SUPL	Superloop parameters
SUPL	vxxx	Add virtual superloops.

Feature operation

The PCA feature operates as outlined below.

Activating and deactivating PCA at the user level

Succession 3.0 introduces three Flexible Feature Codes (FFC) that enable the user to activate, deactivate, or change the target DN on PCA.

To activate or deactivate PCA, perform the following steps

1. Press the DN of any terminal connected to the PBX on which PCA is configured.
2. Enter one of the following FFC codes:
   - **a.** To activate or change PCA, enter PCAA FFC.
   - **b.** To deactivate PCA, enter PCAD FFC.
   - **c.** To verify the current status of PCA, enter PCAV FFC.
3. Enter the prime DN of the desktop.
4. When prompted for a password, enter the Station Control Password (SCPW) of the desktop set or that of PCA (if configured). If no desktop set is configured, the SCPW must be configured on PCA.
5. Enter #, the end-of-dialing digit.
6. Listen for a confirmation tone after entering the main extension number. This tone indicates that the password and extension match and the procedure was successful. If you receive a fast busy tone, the procedure failed and you must hang up and try again.

7 If you are done, enter # to complete the PCAA FFC. Optionally, to update the HOT P DN and activate PCA, enter a new target DN (HOT P DN) followed by #. Listen for a confirmation tone after entering the main extension number. This tone indicates that the password and extension match and the procedure was successful. If you receive a fast busy tone, the procedure failed and you must hang up and try again.

If any of the following events occur, the user hears the overflow tone:

- The SCPL prompt in LD 15 is set to 0 and there is no SCPW configured for the desktop set.
- The user enters the PCA FFCs and the system is not equipped with the PCA package.
- The user enters the PCA FFCs and the LD 15 customer data does not have PCA set to ON.
- The user enters the PCA FFCs for a set that has no PCA in the MADN group.
- The user enters the TPDN FFC for a set that has no PCA in the MADN group.
- The user enters an invalid DN.
- The user enters a DN other than BCS/Ether set DN.
- The user enters “*” or “#” as part of the password.
- The password does not match any SCPWs in the MADN group.

Setting up PCA

The following steps are used in the internal setup:

1 Associate PCA and desktop by sharing a common DN. Configure PCA with the MADN as the primary DN. To change the behavior of the MADN, use multiple and single call ringing. A virtual TN is required for each PCA.

2 Configure Key 0 with the same type of key 0 and DN as the desktop set.

3 Configure Key 1 as a new PCA key (equivalent to a Hot Line Key). If the DN is not configured, the system generates a blending service DN based on the customer setting in LD 15.

Note: Configuration can be simplified through the use of set models.

Phantom Terminal Numbers

Contents

This section contains information on the following topics:

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Feature description

The Phantom Terminal Numbers (PHTN) feature permits system users to define and configure Terminal Numbers (TNs) with no associated physical hardware. Normally, a TN with no associated hardware is disabled.

With Phantom TNs configured, users can define phantom Directory Numbers (DNs) as well. This feature, in conjunction with Call Forward All Calls (CFW) and Remote Call Forward (RCFW), allows a call to a phantom DN to be redirected to an existing telephone.

Phantom TNs can use loops 0-159 for Large Systems. Phantom TNs on Small Systems and Succession 1000 systems are restricted to card slots 61-80 (which convert to Superloops 96-115). The maximum number of Phantom TNs configurable on Small Systems and Succession 1000 systems is 640.

Note: The Phantom Terminal Numbers feature is not to be used for predictive dialing applications. For information on configuring Phantom TNs for predictive dialing, please refer to "Predictive Dialing" on page 347 in this guide.

Operating parameters

Phantom TNs can only have Single Appearance DNs.

All DNs configured on Phantom TNs must conform to the current customer-defined dialing plan.

LD 25 (Move Data Blocks) is not supported between phantom and nonphantom loops; however, it is supported between phantom loops.

Only analog (500/2500 type) telephones support Phantom TNs.

Model telephones (such as TN 500M) are not supported.

A Phantom TN requires one of the phantom terminal loop types shown in Table 21.

Table 21
Supported phantom terminal loop types

Mnemonic	Description
TERM	Single (1) density terminal loop, configured in LD 17.
TERD	Double (2) density terminal loop, configured in LD 17.
TERQ	Quadruple (4) density terminal loop, configured in LD 17.
SUPL	Superloop (8) density terminal loop, configured in LD 97.

Feature interactions

Attendant Administration

This feature is not supported. Phantom DNs cannot be configured on a nonphantom TN.

Attendant Blocking of Directory Number

DNs on Phantom TNs will not be overridden by the Attendant Blocking of DN feature.

Automatic Call Distribution

Phantom TNs cannot be configured as Automatic Call Distribution (ACD) agents.

Call Detail Recording

Call Detail Recording records interact with a Phantom TN exactly the same as with an existing TN with its CFW feature turned on.

Call Forward All Calls

Call Forward All Calls is used in conjunction with RCFW to redirect incoming calls to a Phantom TN/DN to a valid DN.

Call Forward and Busy Status

When a user attempts to define a BFS key for a phantom TN, the system generates the following error message: “An invalid TN has been entered for the Busy/Forward Status (BFS) key.”

Call Forward, Internal Calls

Internal Call Forward cannot be enabled on a phantom TN.

Call Forward/Hunt Override Via Flexible Feature Code

Phantom Terminal Numbers are not overridden by the Call Forward/Hunt Override Via FFC feature. If Call Forward/Hunt Override Via FFC is used against a phantom TN the call will be canceled and overflow tone will be given.

Call Forward, Remote (Attendant and Network Wide)

A Phantom TN does not physically exist, but can be configured with limited hardware (that is, no sets or line cards); however, all required data blocks are configured.

The Phantom TN feature uses the RCFW feature to configure and activate/deactivate the CFW DN on the Phantom TNs.

As the data blocks associated with Phantom TNs match those of standard analog (500/2500 type) telephones configured within the system, the operation of the RCFA and RCFD features on Phantom TNs is applicable to the RCFW feature. As such, the set-based local and network RCFW features can be used to configure and activate/deactivate the CFW DN of Phantom TNs.

The Phantom TN feature uses a Default Call Forward (DCFW) DN. If call forward is not active on the Phantom TN, all calls to the Phantom TN DN are routed to the DCFW DN.

The Phantom TN feature modifies the set-based RCFW feature so that if CFW is not active on the Phantom TN, and the CFW DN entered in the RCFV operation matches the DCFW DN, confirmation tone is returned to the RCFV user; if the CFW DN entered does not match the CDFW DN, overflow is returned.

This change to the set-based RCFV operation is applicable to the network RCFV operation. The operation of this feature network wide requires no changes to the ISDN message passing for the set-based network RCFV operation.

There is no Attendant RCFW operation which interacts with the DCFW DN of Phantom TNs.

Hot Line

Hot Line does not support Phantom TNs.

Call Forward, Internal
Manual Line Service
Multiple Appearance
Multiple Appearance Directory Number Redirection Prime
Station Category Index

These features cannot be enabled on a Phantom TN.

DPNSS1 Diversion

If an incoming call to a Phantom TN contains a DIVERSION BY-PASS REQUEST, Call Forward All Calls applies.

Meridian Link

Phantom TNs cannot be used for origination and termination of calls. AST Class of Service is not allowed on Phantom TNs. With the Telelink Mobility Switch feature, a separate type of TN can be used by Meridian Link AST applications.

Meridian Mail

Phantom DNs are treated like other DNs; a phantom DN can have a Meridian Mail box.

Network Ring Again

The Network Ring Again (NRAG) feature is supported for a Phantom TN with Default Call Forward (DCFW) to an internal set. When the called party becomes idle, the originating caller receives a “set-free” notification. The originating party then presses the Ring Again key, and the DN of the Phantom TN is dialed.

Network Ring Again is not supported for Second Level Default Call Forward or Default Call Forward to an external set.

Override

Call Forward cannot be overridden on phantom terminal numbers. The overflow tone occurs if an Override is attempted.

Recorded Announcement for Calls Diverted to External Trunks

If a Phantom TN is forwarded to an external outgoing CO route and the Recorded Announcement for Calls Diverted to External Trunks feature is configured for this route, the calling party that is forwarded due to the Phantom TN feature will be provided with a recorded announcement.

Remote Call Forward

If Remote Call Forward is to be used in conjunction with Phantom TNs, then the Phantom TNs must be configured with the Call Forward All Calls (CFW) feature.

Ring Again on No Answer

Although Ring Again on No Answer can be applied to a phantom DN, it is not recommended. Because a phantom DN cannot be active or busy, the caller is not notified when the phantom DN's forward DN does not answer.

Secretarial Filtering

If a Phantom TN is call forwarded to an existing telephone, and that telephone is used to call the DN on the Phantom TN, the call receives DCFW treatment.

Set-Based Administration Enhancements

Set-Based Administration supports making changes to Phantom TNs with the exception of changing Hunt DNs, since Phantom TNs cannot have Hunt DNs.

Virtual Office

Virtual Terminal Numbers and Phantom Terminal Numbers use the same TN range. The maximum number of Phantom TNs and Virtual TNs configurable on Small Systems and Succession 1000 systems is 640.

Feature packaging

The Phantom Terminal Numbers (PHTN) feature is available as package 254.

Using Remote Call Forwarding (RCFW) with Phantom TNs requires Flexible Feature Codes (FFC) package 139.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure a phantom loop.

2 LD 97 – Configure a phantom Superloop.

3 LD 10 – Define a TN for the phantom loop.

LD 17 – Configure a phantom loop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Common Equipment parameters.
- TERM	N0-N159	Single density local terminal loop; precede loop number with “N” to create a phantom loop; precede with an “X” to remove a terminal loop.
- TERD	N0-N159	Double density local terminal loop; precede loop number with “N” to create a phantom loop; precede with an “X” to remove a terminal loop.
- TERQ	N0-N159	Quadruple density local terminal loop; precede loop number with “N” to create a phantom loop; precede with an “X” to remove a terminal loop.

LD 97 – Configure a phantom Superloop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop data.
SUPL	N0-N156	Local Superloop in multiples of four
	N96-N115	For Succession 1000M Cabinet and Meridian 1 Option 11C Cabinet systems Precede loop number with “N” to create a phantom loop; precede with an “X” to remove a terminal loop. ***Note:*** Phantom TNs can use loops 0-159 for Large Systems. Phantom TNs on Small Systems and Succession 1000 systems are restricted to card slots 61-80 (which convert to superloops 96-115).

LD 10 – Define a TN for the phantom loop.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	500	Telephone type.
TN	 l s c u c u	Terminal Number; if the loop is a phantom loop, "PHANTOM" is echoed to the technician. For Large Systems For Small Systems and Succession 1000 systems
DN	xxx...x	Directory Number; must be a Single Appearance DN
CLS	aaaa	Class of Service options, which cannot include AGTA, CCSA, MNL, or LPA.
FTR	DCFW ll xxx...x	Default DCFW length (ll) and default CFW DN xxx...x (up to 23 digits).

Feature operation

Operation of this feature with Call Forwarding is described below.

1 A call is directed to a phantom DN.

2 If the phantom DN is Call Forward Activated, the call is directed to its CFW DN.

3 If the phantom DN is Call Forward Deactivated, the call is directed to its Default CFW DN.

Position Busy with Call on Hold

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Feature description

This feature prevents an attendant from going into Position Bushy when a call on a Loop Key is on hold, or the source or destination of an active loop key is excluded from the call.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Forward No Answer

If an attendant with a call on hold does not answer an Attendant Forward No Answer call within a customer-defined time, the console is not placed in Position Busy.

Scheduled Access Restriction

If an attendant in a Scheduled Access Restriction group has a call on hold, the attendant is not placed in Position Busy when the group enters an off-hour period.

Feature packaging

This feature is packaged under International Supplementary Features (SUPP), package 131.

Feature implementation

LD 15 – Configure Position Busy with Call on Hold.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options.
...		
- OPT	(BOHA) BOHD	Position Busy with Calls on Hold (allowed) denied.

Feature operation

With Position Busy with Calls on Hold Allowed, (BOHA) configured in LD 15, normal operation is not changed when an attendant with a call on hold presses the POS BUSY key. The attendant goes into Position Busy.

With Position Busy with Calls on Hold Denied (BOHD) configured in LD 15, when an attendant with a call on hold presses the POS BUSY key the system will react as if nothing has happened.

In addition, if the attendant with a call on hold presses the **POS BUSY** key, the system remains in day service (even if supposed to go in Night Service).

Predictive Dialing

Contents

This section contains information on the following topics:

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Feature description

With Predictive Dialing, the process of making outgoing calls to customers is automated for Automatic Call Distribution (ACD) agents. Host applications can request the system to make calls using autodialers or phantom TNs. When a call is answered, the application sends a request to the switch to transfer the call to a live agent. The call needs to be transferred before, or while, the customer starts speaking in order to prevent customers from abandoning the call if they think no one has answered them. This transfer was previously performed by Meridian Link in two steps by sending two separate Application Module Link (AML) messages to initiate and then complete the transfer. This operation takes a minimum of 400 to 450 milliseconds for the system to process.

The Fast Transfer feature 21 allows applications residing on the Application Module (AM) or host computers to transfer a call in one step (a blind transfer) by sending only one AML message (Fast Transfer) to the switch, thereby saving approximately 200 to 250 milliseconds of transfer time. This Fast Transfer feature is useful for predictive applications to make outbound calls and then quickly transfer them once the customer has answered (that is, live voice has been detected). Fast Transfer can also be used in a non-predictive dialing environment. Applications that want to perform a blind transfer can now execute it more quickly.

The Predictive Dialing feature enables applications residing on the AM or host computers to send a combined Make Call and Transfer request on behalf of an autodialer or Phantom TN. As soon as live voice is detected by third-party equipment, or notification is sent to the switch indicating the call has been answered (for example, answer supervision), the application can send the Fast Transfer request to the switch immediately transferring the call to an ACD agent.

Operating parameters

When Phantom TNs/DNs are used to originate calls as part of a predictive dialing operation, Small Systems and Succession 1000 systems are not supported.

Attendant Consoles, and Basic Rate Interface sets cannot initiate Fast Transfer or predictive calls.

The system does not support live voice answer detection. Live voice answer detection is currently achieved through third-party vendor equipment.

If phantom TNs/DNs are used, this development only supports calls and Fast Transfers originated by phantom TNs/DNs which are defined as Associate set (AST) Meridian 1 proprietary telephones on a phantom loop.

Data calls are not supported.

For outbound trunk calls, if no third-party equipment is used to detect live voice answer, the switch will have to depend on receiving answer supervision before transferring the call to the target DN.

If voice detection is used, the application will not be able to Fast Transfer the call before the call is established (that is, answer notification is received).

The application will not be able to complete the transfer when Fast Transferring over a trunk.

Not all analog trunks support answer supervision. Not all digital trunks provide answer supervision. For trunks that do not support answer supervision, the End-of-Dialing (EOD) timer will be used to trigger the transfer.

Receiving answer supervision depends on the accuracy of signals returned by the external network. Answer supervision may be received before an EOD timeout, pseudo answer supervision may also be received due to an EOD timeout. A pseudo answer supervision may be received if the far-end has an EOD timeout even though the local switch has answer supervision configured.

The AML requires an Enhanced Serial Data Interface (ESDI) card or Multi-purpose Serial Data Link (MSDL) card (NT6D80AA) on the switch. If a Small System or Succession 1000 system is used, a Serial Data Interface/D-Channel (SDI/DCH) card (NTAK02AA) is required to configure the ESDI port.

The AML connection requires an RS232 cable.

Meridian Link software is required for host applications to utilize this feature.

Feature interactions

Call Hold, Deluxe
Call Hold, Permanent

If an established call is put on hold by the set initiating the Fast Transfer, the switch will not be able to transfer the call. The switch can only transfer a call if it is in the established state.

Call Transfer by Meridian 1 proprietary telephone

The application sends the Fast Transfer request on behalf of a Meridian 1 proprietary telephone, and then the switch initiates and completes the transfer immediately which is similar to a normal call transfer from a Meridian 1 proprietary telephone.

In a Predictive Dialing scenario where the autodialer (originating DN) is a Meridian 1 proprietary telephone, the Make Call message sent by the application to the switch to make a call on behalf of the Meridian 1 proprietary telephone, and then the call transfer call, will interact with the Meridian 1 proprietary telephone Call Transfer feature. The autodialer is configured with Class of Service TRN so that the switch can transfer the call to the target destination.

Call Transfer by Analog (500/2500 type) Telephone

The application sends the Fast Transfer request on behalf of an analog (500/2500 type) telephone. The switch will then initiate and complete the transfer in one step.

In a predictive dialing scenario, the application will send the Make Call request on behalf of the autodialer (analog (500/2500 type) telephone) to have the switch make the call, and then transfer the call when the switch receives the Fast Transfer message. The autodialer needs to be configured with Classes of Service Dial Pulse (DIP) and Transfer Allowed (XFA) for 500 sets, or with Classes of Service Digitone (DTN) and XFA for 2500 sets.

Command and Status Link

The Command and Status Link, also known as the AML, is the link on which the messages for the Predictive Dialing feature flow between the switch and an Application Module. The CON/FastTransfer is an AML message.

Trunks

Only certain trunks will support answer supervision. The End-of-Dialing timer will be used for trunks that do not support answer supervision.

Feature packaging

There are no new software packages required for the Predictive Dialing feature. However, the following packages are required to utilize the feature:

- Application Module Link (IAP3P) package 153, and
- Meridian Link Module (MLM) package 209 if the Meridian Link Module is involved.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure the ESDI port to the Meridian Link Module.

2 LD 17 – Configure the MSDL port to the Meridian Link Module.

3 LD 10 – Configure non-ACD analog (500/2500 type) telephones as autodialers.

4 LD 11 – Configure non-ACD Meridian 1 proprietary telephones as autodialers.

5 LD 23 – Configure ACD groups.

6 LD 10 – Configure ACD analog (500/2500 type) telephones as autodialers.

7 LD 11 – Configure ACD Meridian 1 proprietary telephones as autodialers.

8 LD 23 – Configure a Control DN (CDN – default mode). If the application wants to transfer a call to a target CDN, a CDN must be configured. CDNs can be in default or controlled mode.

9 LD 23 – Configure a Control DN (CDN – controlled mode).When a CDN is in controlled mode, the application can have control of the call once it enters the CDN.

10 LD 14 – Define answer supervision for trunks. If the application wants to transfer outgoing calls based on answer supervision, answer supervision must be configured. If answer supervision is not configured, the End-of-Dialing timer will be used as a trigger for the system to transfer the call.

11 LD 16 – If the application is using the End-of-Dialing timer to transfer outbound calls, the timer must be configured in the Route Data Block.

12 LD 17 – In order to originate calls from phantom TNS/DNs, a phantom loop must first be configured and a physical loop card must be installed. A phantom DN can then be configured as part of a specific device group. After configuration changes to the loop card, the system must be reinitialized for the changes to take effect.

13 LD 97 – If a superloop is used, the phantom loop is configured in this overlay.

14 LD 11 – After configuring the phantom loop, an AST Meridian 1 proprietary set can be designated to a specific device group which can be controlled by applications. Therefore, when an application wants to originate a call on behalf of an idle TN, it can use a phantom TN. This idle TN is an AST Meridian 1 proprietary set which is defined on a phantom loop. There is no upper limit on the number of devices per group defined by the Phantom DN. However, there is an upper limit on the number of TNs that can be defined for the loop card. This number is dependent on the density of the loop card. The ITNA and DGRP prompts must be configured as follows:

This feature does not require any changes to the overlays. The following illustrates the configuration requirements to set up this feature. Most of these requirements are used by existing Meridian Link and Application Module applications.

LD 17 – Configure the ESDI port to the Meridian Link Module.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	All input/output devices (includes D-Channels)
- CTYP	ESDI	Card Type. ESDI card.
- DNUM	x	Device number is x.
- DES	NEWTTY	Description of this I/O device.
- BPS	19200	Baud rate is 19,200 bits per second.
- CLOK	INT	Internal clocking.

- IADR	3	HDLC protocol individual address.
- RADR	1	HDLC protocol remote address.
TYPE	PARM	System parameters
...		
- CSQI	(20)	Maximum call registers for Command and Status Link (CSL) input queues (use the default, unless the system requires otherwise).
- CSQO	(20)	Maximum call registers for CSL output queues (use the default, unless the system requires otherwise).
TYPE	VAS	Value Added Server
...		
- VSID	y	Server ID y.
- AML	x	Port used by AML defined earlier in this overlay.
- - SECU	YES	Security on for Meridian Link.
- - INTL	x	Length of time interval (five-second increments) (for example, 2).
- - MCNT	x	Threshold for number of messages per time interval (for example, 100). MCNT must be reduced to 300 if CCR and/or Meridian Link are used on Large Systems.
- - CONF	DIR	Direct link configuration.

LD 17 – Configure the MSDL port to the Meridian Link Module.

Prompt	**Response**	**Description**
REQ	CHG	Change.
TYPE	ADAN	All input/Output devices (includes D-Channels)
...		

- CTYP	MSDL	Card Type. MSDL card.
- DNUM	y	Device number is y. Refers to the device number on the MSDL card.
- DES	MERIDIAN_LINK	Description of this I/O device.
- BPS	19200	Baud rate is 19,200 bits per second.
- PARM	RS232 DCE	Parameters for interface and transmission mode. DTE/DCE setting.
- IADR	3	HDLC protocol individual address.
- RADR	1	HDLC protocol remote address.
TYPE	PARM	Gate opener.
...		
- CSQI	(20)	Maximum call registers for CSL input queues (use the default, unless the system requires otherwise).
- CSQO	(20)	Maximum call registers for CSL output queues (use the default, unless the system requires otherwise).
TYPE	VAS	Value Added Server
...		
- VSID	y	Server ID y.
- AML	x	Port used by AML x, defined earlier in this overlay.
- - SECU	YES	Security on for Meridian Link.
- - INTL	x	Length of time interval (five-second increments) (for example, 2).
- - MCNT	x	Threshold for number of messages per time interval (for example, 100). MCNT must be reduced to 300 if CCR and/or Meridian Link are used on Large Systems.
- - CONF	DIR	Direct link configuration.

LD 10 – Configure non-ACD analog (500/2500 type) telephones as autodialers.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Telephone type.
...		
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CUST	xx	Customer number, as defined in LD 15
...		
DN	x...x	Internal Directory Number.
AST	YES	Associate set assignment. The internal DN is an AST.
CLS	XFA	Transfer allowed.
CLS	DIP	Dial Pulse Class of Service for 500 sets (use DTN for 2500 sets).

LD 11 – Configure non-ACD Meridian 1 proprietary telephones as autodialers.

Prompt	Response	Description
REQ	NEW	New.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

...		
CUST	xx	Customer number, as defined in LD 15
...		
KLS	1-7	Number of key lamp strips, typically one.
...		
AST	xx yy	Key number for Associate set DN assignment.
...		
KEY	xx SCR yyyy	Key number, Single Call Ringing, DN.
KEY	xx TRN	Key number, Call Transfer.
KEY	xx AO6	Key number, six-party conference.
KEY	xx SCR yyyy	Key number, Single Call Ringing, second DN.
CLS	xx RLS	Key number, Release.

LD 23 – Configure ACD groups.

Prompt	Response	Description
REQ	NEW	New.
TYPE	ACD	Automatic Call Distribution data block.
CUST	xx	Customer number, as defined in LD 15
ACDN	xxxx	ACD Directory Number.
...		
ISAP	YES	Integrated Services Application Protocol. ACD DN uses Meridian Link (ISDN/AP) messaging.
- VSID	0-15	Value Added Server ID. This Server ID used for Meridian Link messaging must match the VSID defined in LD 17.

LD 10 – Configure ACD analog (500/2500 type) telephones as autodialers.

Prompt	Response	Description
REQ	NEW	New.
TYPE	500	Telephone type.
...		
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
...		
DN	x...x	Internal Directory Number.
AST	YES	Associate set assignment. The internal DN is an AST.
...		
CLS	AGTA	ACD agent allowed Class of Service.
CLS	DIP	Dial Pulse Class of Service for 500 sets (use DTN for 2500 sets).
...		
AACD	YES	ACD telephone is an Associate set.
FTR	ACD xxxx yyyy	ACD DN and the ACD position ID.

LD 11 – Configure ACD Meridian 1 proprietary telephones as autodialers.

Prompt	Response	Description
REQ	NEW	New.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CUST	xx	Customer number, as defined in LD 15
...		
KLS	1-7	Number of key lamp strips, typically one.
...		
AST	xx yy	Key numbers for Associate set DN assignment.
...		
KEY	0 ACD xxxx yyyy	Key 0, ACD, ACD DN, and agent's ID.
KEY	xx MSB	Key number, Make Set Busy.
KEY	xx NRD	Key number, Not Ready.
KEY	xx TRN	Key number, Call Transfer.
KEY	xx AO6	Key number, six-party conference.
KEY	xx SCR yyyy	Key number, Single Call Ringing, second DN.
CLS	xx RLS	Key number, Release.

LD 23 – Configure a Control DN (CDN – default mode). If the application wants to transfer a call to a target CDN, a CDN must be configured. CDNs can be in default or controlled mode.

Prompt	Response	Description
REQ	NEW	New.
TYPE	CDN	Control Directory Number data block.
CUST	xx	Customer number, as defined in LD 15
CDN	xxxx	DN of the Control DN (counts as an ACD DN).
...		
DFDN	xxx...x	Default destination ACD DN.
CEIL	0-(2047)	CDN ceiling value. CEIL limits the number of unanswered calls a CDN can have at its default ACD DN at a time. Enter the maximum value (the default).
...		
RPRT	YES	Report Control.
CNTL	NO	NO sends CDN calls to the Default ACD DN.

LD 23 – Configure a Control DN (CDN – controlled mode).When a CDN is in controlled mode, the application can have control of the call once it enters the CDN.

Prompt	Response	Description
REQ	NEW	New.
TYPE	CDN	Control Directory Number data block.
CUST	xx	Customer number, as defined in LD 15
CDN	xxxx	DN of the Control DN (counts as an ACD DN).
...		
DFDN	xxx...x	Default destination ACD DN.
CEIL	0-(2047)	CDN ceiling value. CEIL limits the number of unanswered calls a CDN can have at its default ACD DN at a time. Enter the maximum value (the default).
...		
RPRT	YES	Report Control.
CNTL	YES	Control DN is in control (the default).
VSID	0-15	Value Added Server ID. Server ID used for Meridian Link messaging (defined in LD 17).
HSID	0-15	Host Link ID used when Customer Controlled Routing and Meridian Link applications are both running.

LD 14 – Define answer supervision for trunks. If the application wants to transfer outgoing calls based on answer supervision, answer supervision must be configured. If answer supervision is not configured, the End-of-Dialing timer will be used as a trigger for the system to transfer the call.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type where: aaa = CAA, CAM, COT, CSA, DID, FEX, FGDT, IDA, TIE, or WAT.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
SUPN	YES	Answer and disconnect supervision are required.

LD 16 – If the application is using the End-of-Dialing timer to transfer outbound calls, the timer must be configured in the Route Data Block.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block
...		
CNTL	YES	Change controls or timers.
- TIMR	EOD 128-(13952)-32640	End-of-Dialing timer in milliseconds. The default is 13952 milliseconds.

LD 17 – In order to originate calls from phantom TNS/DNs, a phantom loop must first be configured and a physical loop card must be installed. A phantom DN can then be configured as part of a specific device group. After configuration changes to the loop card, the system must be reinitialized for the changes to take effect.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CEQU	Common Equipment parameters
...		
- TERM	0-159 [X] 0-159 [C] 0-159	Single density local terminal loops. Precede loop number with X to remove. Precede loop number with C to create a phantom loop.
- TERD	0-159 [X] 0-159 [C] 0-159	Double density local terminal loops. Precede loop number with X to remove. Precede loop number with C to create a phantom loop.
- TERQ	0-159 [X] 0-159 [C] 0-159	Quad density local terminal loops. Precede loop number with X to remove. Precede loop number with C to create a phantom loop.

LD 97 – If a superloop is used, the phantom loop is configured in this overlay.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop parameters.
SUPL	0-156 [X] 0-156 [C] 0-156	Superloop number in multiples of four. Precede superloop number with X to remove. Precede superloop number with C to create a phantom superloop.

LD 11 – After configuring the phantom loop, an AST Meridian 1 proprietary set can be designated to a specific device group which can be controlled by applications. Therefore, when an application wants to originate a call on behalf of an idle TN, it can use a phantom TN. This idle TN is an AST Meridian 1 proprietary set which is defined on a phantom loop. There is no upper limit on the number of devices per group defined by the Phantom DN. However, there is an upper limit on the number of TNs that can be defined for the loop card. This number is dependent on the density of the loop card. The ITNA and DGRP prompts must be configured as follows:

Prompt	Response	Description
REQ:	NEW	New.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CDEN	 SD DD 4D	Card density. Single density. Double density. Quad density.
DES	phanDN	One-to-six character Office Data Administration System (ODAS) Station Designator.
CUST	xx	Customer number, as defined in LD 15
...		
CLS	NDD	No digit display is recommended if configuring phantom devices.
CLS	(DNDD)	Dialed Name Display denied is recommended if configuring phantom devices.
...		
AST	00	Key 0 is AST.

IAPG	(0)-15	Meridian Link Unsolicited Status Message (USM) group. These groups determine which status messages are sent for an AST set. The default 0 sends no messages, whereas Group 1 sends all messages.
ITNA	(NO) YES	Idle TN for Third Party Application. Set ITNA to YES for Phantom TN calls.
DGRP	(1)-5	Device Group with which phantom TNs are associated.
...		
KEY	xx SCR yyyy	Key number, Single Call Ringing, DN.
CLS	xx RLS	Key number, Release.

Feature operation

Applications invoke the Fast Transfer feature by sending a Fast Transfer request message to the switch. No specific operating instructions are required to use this feature.

Pretranslation and System Speed Call Enhancement

Contents

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Feature description

Pretranslation and System Speed Call Enhancement provides the option to allow or deny Pretranslation when a System Speed Call list entry is dial accessed.

The existing Pretranslation feature allows the creation of a flexible dialing plan by using Speed Call lists which are modified for pretranslation. The dialing capabilities and/or restrictions of each Pretranslation group are defined in Pretranslation Tables.

The existing System Speed Call feature allows abbreviated dialing and also allows users to temporarily override the set's Class of Service, Trunk Group Access Restrictions (TGARs), and Code Restrictions.

Analog (500/2500 type) sets, Meridian 1 proprietary sets, and Attendant Consoles can activate System Speed Call by using a Special Prefix (SPRE) or Flexible Feature Code (FFC).

For further information pertaining to the existing Pretranslation and System Speed Call features, refer to the feature modules in this guide.

Table 22 and Table 23 are examples of a Pretranslation Table and a System Speed Call list respectively.

Table 22
Example of Pretranslation Table

List entry	Corresponding DN or Code	Function
0	space <CR>	Pass Pretranslation digit unchanged
1	space <CR>	Pass Pretranslation digit unchanged
2	space <CR>	Pass Pretranslation digit unchanged
3	space <CR>	Pass Pretranslation digit unchanged
4	space <CR>	Pass Pretranslation digit unchanged
5	space <CR>	Pass Pretranslation digit unchanged
6	space <CR>	Pass Pretranslation digit unchanged
7	8000	Convert to Route Access Code 8000
8	***	Delete Pretranslation (first dialed) digit, pass remaining digits unchanged
9	*	Block the call

Table 23
Example of System Speed Call List

List entry	Corresponding DN
00	7182
01	122455678
...	...

In Table 22, if the first dialed digit is 0 to 6, Pretranslation passes all of the digits and leaves them unchanged. If the first dialed digit is 7, Pretranslation changes digit 7 to Route Access Code 8000. If the first dialed digit is 8, Pretranslation deletes the first dialed digit and passes the remaining digits unchanged. If the first dialed digit is 9, Pretranslation blocks the call.

To dial access System Speed Call lists, the user dials:

1 SPRE, as defined in LD 15

2 System Speed Call Feature Code - 73

3 System Speed Call list entry number

If the system is equipped with Flexible Feature Codes, the user dials:

1 FFC, as defined in LD 57.

2 System Speed Call list entry number

With the existing Pretranslation and System Speed Call features, when Dial Access occurs, Pretranslation is performed on the first dialed digit of the Special Prefix (SPRE) or Flexible Feature Code (FFC). The first digit of the digits stored in the System Speed Call list entry is then also pretranslated.

The Pretranslation and System Speed Call Enhancement introduces the BPSS prompt in LD 15. This prompt provides the option to allow or deny pretranslation on the System Speed Call list entry when dial accessed. If BPSS is set to YES in LD 15, Pretranslation is blocked. Therefore, only the first dialed digit is pretranslated. The first digit of the digits stored in the System Speed Call list entry is not pretranslated.

To follow are examples of Pretranslation and System Speed Call functionalities when Pretranslation is blocked/not blocked. Table 22 and Table 23 are considered for these examples. It is assumed that the SPRE method of dialing is used and that the user has the following configuration:

- Special Prefix (SPRE) code - **1**
- System Speed Call Feature Code - **73**

BPSS = NO

With dial access and the BPSS option set to NO in LD 15, Pretranslation is not blocked. Therefore, the existing Pretranslation functionality is retained.

When the user dials 1+73+00, Pretranslation occurs twice. It occurs once on the first dialed digit (1) and once again on the first digit of the digits stored in the System Speed Call list entry (7 of 7182).

When the user dials SPRE + 73 + 00, the first digit of the digits stored in the System Speed Call list entry (7 of 7182) is converted to DN 8000. In this example, DN 8000 is a Trunk Route Access Code; therefore, the call goes out on that route, and the digits 182 are outpulsed.

BPSS = YES

With dial access and the BPSS option set to YES in LD 15, Pretranslation is blocked. Therefore, the new Pretranslation functionality is in effect.

When the user dials 1+73+00, the first dialed digit (1) is pretranslated. However, the first digit of the digits stored in the System Speed Call list entry (7 of 7182) is not pretranslated.

When the user dials SPRE + 73 + 00, the list entry number is converted to DN 7182. When BPSS = YES, Pretranslation is blocked at this point. Therefore, the first digit of the digits stored in the System Speed Call list entry (7 of 7182) is not converted to the corresponding DN (8000) in the Pretranslation Table.

Operating parameters

To allow or deny Pretranslation on a System Speed Call list entry when dial accessed, the BPSS prompt must be defined in the Customer Data Block.

When Pretranslation is disabled (PREO = 0) in the Customer Data Block, BPSS is prompted but does not take effect. Therefore, the current functionality is retained.

With Dial Access and the BPSS option set to YES in the Customer Data Block, only the first dialed digit is pretranslated. The first digit of the digits stored in the System Speed Call list entry are not pretranslated.

With Dial Access and the BPSS option set to NO in the Customer Data Block, the existing operation is retained.

The functionality of the Speed Call (Dial Access and Key Access) and System Speed Call (Key Access only) features is not changed by this enhancement.

The operation of Key Access to System Speed Call with Pretranslation is not modified with this feature.

Existing dialing plans are affected when the Pretranslation and System Speed Call Enhancement is configured.

The pre-programmed DN in the System Speed Call list can be internal or external to the system.

Feature interactions

There are no new feature interactions as a result of this enhancement. Refer to the “Pretranslation” on page 373, “Speed Call” on page 749, and “Speed Call, System” on page 759 in this book for a list of existing feature interactions.

Feature packaging

The following packages are required for Pretranslation and System Speed Call Enhancement:

- System Speed Call (SSC) package 34
- Pretranslation (PXLT) package 92

Feature implementation

Note: The Pretranslation and System Speed Call features must be configured as per the existing implementation procedures. Refer to the Pretranslation feature module and the System Speed Call feature module in this guide.

CAUTION

Care must be taken when implementing Pretranslation and System Speed Call Enhancement, as existing dialing plans will be impacted when BPSS = YES. In this case, the existing Pretranslation functionality is changed, and the entire Customer group of dial access System Speed Call users is affected.

LD 15 – Allow or deny blocking of Pretranslation on list entry when dial accessed.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		

PREO	1	Pretranslation Option enabled. 0 = Pretranslation Option disabled (default).
BPSS	YES	Block Pretranslation on System Speed Call lists when dial accessed. NO = Do not block Pretranslation on System Speed Call lists when dial accessed (default).

Feature operation

To dial access System Speed Call lists, the user

1 Lifts the handset of the analog (500/2500 type) set, Meridian 1 proprietary set, or Attendant Console

2 Dials the Special Prefix (SPRE) code, as defined in LD 15

3 Dials the System Speed Call Feature Code - **73**

4 Dials the System Speed Call list entry number

If the system is equipped with Flexible Feature Codes (FFCs), the user

1 Lifts the handset of the (500/2500 type) set, Meridian 1 proprietary set, or Attendant Console

2 Dials the Flexible Feature Code (FFC) for accessing System Speed Call, as defined in LD 57

3 Dials the System Speed Call list entry number

Pretranslation

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Feature description

In a business or hospitality environment, many communications situations can be simplified with a flexible dialing plan. Pretranslation lets you create such a plan by using Speed Call lists as Pretranslation Tables.

Some typical uses of Pretranslation are:

- room number to DN correlation
- partitioning of telephones by category, group, department, floor, building, room, or special service
- internal call restrictions, and
- expanded customer dialing capability.

The dialing capabilities and/or restrictions of each Pretranslation group are defined in Pretranslation Tables. The tables are Speed Call lists modified for Pretranslation.

With Pretranslation, only the first dialed digit of a call is pretranslated. The translation choices are:

- **Pass** the digit as dialed with no changes
- **Replace** the first dialed digit with a specified substitute digit or digits, and pass the remaining digits unchanged
- **Delete** the first dialed digit and pass the remaining digits unchanged, or
- **Block** the call based on the first digit dialed.

The pretranslator must deal with all telephones, trunks, and consoles capable of delivering a dialed digit to the system digit processor. Each of these must be assigned to one of 255 Pretranslation groups.The groups are generally set up as follows:

- trunk and Direct Inward System Access (DISA) calls default to group 0
- Attendant Consoles default to group 1, and
- telephones and terminals default to group 0, but may be assigned to groups 2-254.

  Note: When Pretranslation group 0 is configured, all sets are affected, as the XLST prompt in LDs 10 and 11 has a default value of 0. The XLST prompt associates a set with a specified Pretranslation group.

The dialing capabilities of each group are reflected by the codes stored against entries in the Pretranslation Table. The four possible codes are shown in Table 24.

Table 24
Pretranslation Table

Code	Function
*	Block call.
***	Delete Pretranslation (first dialed) digit, pass remaining digits unchanged.
space <CR>	Pass Pretranslation digit unchanged.
xxxx...x	Pretranslate digit into xxxx...x, where: xxxx...x = replacement DN.

Only the first dialed digit is sent from the digit processor to the pretranslator. The pretranslator looks up the stored code for the dialed digit in the Pretranslation Table associated with the calling terminal, applies the treatment specified by the entry, and passes the result to the DN translator. From then on, the call is processed normally. Pretranslation of the call is finished at this point, unless call modification procedures, such as a Call Transfer, are involved.

Setting up dialing plans and Pretranslation Tables

Steps needed to set up Pretranslation:

1 Identify the customer numbering plan.

2 Determine access and restrictions for each Pretranslation calling group.

3 Determine dialing requirements and instructions for the Pretranslation calling groups and create a Pretranslation Table for each group.

4 Implement the feature.

A hotel has been chosen as a model to illustrate the principles of Pretranslation and how to set up Pretranslation. However, Pretranslation can be applied to many other business environments.

Table 25
Description of Pretranslation model

Hotel with 12 floors containing administrative offices, hotel services, and guest rooms.
Floor 1 – Lobby, gift shop, restaurants, and administrative offices.
Floor 2 – Meeting rooms, salon, and additional office space.
Floor 3 – Banquet rooms and health club.
Floors 4-12 – Guest rooms (floors 4-9 each have 50 rooms, floors 10-12 each have 25 suites).

Step 1 – Identify the numbering plan

The model hotel's numbering plan is shown in Table 26.

Table 26
Numbering plan for model

Available numbers	Assigned to	Actual DNs used
0	Operator	0
1	Guest rooms on floor 10	1001-1026
	Guest rooms on floor 11	1101-1126
	Guest rooms on floor 12	1201-1226
2	Room service	2001
	Cafe	2002
	Restaurant	2003
	Gift shop	2004
	Health club	2005
	Salon	2006
	Housekeeping	2007
	Bell Captain	2008
	Valet	2009
	Meeting rooms	2100-2199
	Administrative offices	2300-2599
	Security	2700
	Front desk	2730
	Lobby telephones	2750-2765
	Miscellaneous	2800-2899
3	SPRE code	
4	unused	
5	unused	
6	Trunk access codes	620-635
7	Guest rooms on floor 4	7401-7451
	Guest rooms on floor 5	7501-7551
	Guest rooms on floor 6	7601-7651
	Guest rooms on floor 7	7701-7751
	Guest rooms on floor 8	7801-7851
	Guest rooms on floor 9	7901-7951
8	unused	
9	BARS access codes	9

Step 2 – Determine access restrictions

Pretranslation calling groups and dialing restrictions are shown in Table 27.

Table 27
Access and restrictions for model

Group number (XLST)	Type of station	Allowed access	Denied access
0	Default for DISA trunks and telephones	Operator only	All except Operator
1	Guest rooms	Other guest rooms, hotel services, local and long distance, operator	Administrative telephones and direct trunk access
2	Lobby and courtesy telephones	Guest rooms, security, and the operator	Hotel services, administrative telephones, local and long distance, direct trunk access, and SPRE
3	Administrative A	Guest rooms, administrative telephones, direct trunk access, SPRE, operator, BARS access for local and long distance	Direct trunk access
4	Administrative B	Guest rooms, administrative telephones, SPRE, operator	Direct trunk access, BARS access for local and long distance

Step 3 – Determine dialing requirements and create Pretranslation Tables

Dialing instructions for Group 0 (zero) in this model are shown in Table 28 and the corresponding Pretranslation Table is listed in Table 29. For an explanation of the groups used in this model, see Table 27.

Table 28
Group 0 – Default for unassigned trunks and telephones

Actual digits dialed	Desired destination
1	Operator
2	Operator
3	Operator
4	Operator
5	Operator
6	Operator
7	Operator
8	Operator
9	Operator
0	Operator

Table 29
Group 0 – Pretranslation Table (default)

Digit	Code	Function	Destination
1	0	replace	Operator
2	0	replace	Operator
3	0	replace	Operator
4	0	replace	Operator
5	0	replace	Operator
6	0	replace	Operator
7	0	replace	Operator
8	0	replace	Operator
9	0	replace	Operator
0	space <CR>	pass	Operator

Dialing instructions for Group 1 in this model are shown in Table 30 and the corresponding Pretranslation Table is listed in Table 31.

Table 30
Group 1 – Guest dialing instructions for model

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2	Security
3	SPRE (housekeeping staff for Room Status)
4	Front desk
51	Room Service
52	Cafe
53	Restaurant
54	Gift shop
55	Health club
56	Salon
57	Housekeeping
58	Bell captain
59	Valet
7xxx	Guest rooms on floors 4-9
8	Long distance calls
9	Local calls
0	Operator

Table 31
Group 1 – Pretranslation Table (Guests)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	2700	replace	Security
3	space <CR>	pass	SPRE
4	2730	replace	Front desk
5 (see Note)	200	replace	Guest services
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	620	replace	Long distance calls
9	space <CR>	pass	Local calls
0	space <CR>	pass	Operator

Note: When a guest dials 51 for room service, the digit "5" is translated to the entry "200" and the 1 is passed as is, resulting in the extension "2001."

Dialing instructions for Group 2 in this model are shown in Table 32 and the corresponding Pretranslation Table is listed in Table 33.

For an explanation of the groups used in this model, see Table 27.

Table 32
Group 2 – Lobby and courtesy telephone dialing instructions

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2	Security
7xxx	Guest rooms on floors 4-9
0	Operator

Table 33
Group 2 – Pretranslation Table (lobby and courtesy telephones)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	2700	replace	Security
3	*	block call	Not used
4	*	block call	Not used
5	*	block call	Not used
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	*	block call	Not used
9	*	block call	Not used
0	space <CR>	pass	Operator

Dialing instructions for Group 3 in this model are shown in Table 34 and the corresponding Pretranslation Table is listed in Table 35.

For an explanation of the groups used in this model, see Table 27.

Table 34
Group 3 – Administrative A dialing instructions for model

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2xxx	Administrative telephones
3	SPRE
7xxx	Guest rooms on floors 4-9
9	Local/long distance through BARS
0	Operator

Table 35
Group 3 – Pretranslation Table (Administrative A)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	space <CR>	pass	Administrative telephones
3	space <CR>	pass	SPRE
4	*	block call	Not used
5	*	block call	Not used
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	*	block call	Not used
9	space <CR>	pass	Local/long distance through BARS
0	space <CR>	pass	Operator

Dialing instructions for Group 4 in this model are shown in Table 36 and the corresponding Pretranslation Table is listed in Table 37.

For an explanation of the groups used in this model, see Table 27.

Table 36
Group 4 – Administrative B dialing instructions for model

Actual digits dialed	Desired destination
1xxx	Guest rooms on floors 10-12
2xxx	Administrative telephones
3	SPRE
7xxx	Guest rooms on floors 4-9
0	Operator

Table 37
Group 4 – Pretranslation Table (Administrative B)

Digit	Code	Function	Destination
1	space <CR>	pass	Guest rooms
2	space <CR>	pass	Administrative telephones
3	space <CR>	pass	SPRE
4	*	block call	Not used
5	*	block call	Not used
6	*	block call	Not used
7	space <CR>	pass	Guest rooms
8	*	block call	Not used
9	*	block call	Not used
0	space <CR>	pass	Operator

Operating parameters

Pretranslation Table codes are limited to the four described on page 375.

User groups are limited to 255 .

Each Pretranslation Table entry can be up to 31 characters long; however, it is recommended that a maximum of eight characters be used.

After Pretranslation, any previously loaded (but not pretranslated) digits are added to the end of the pretranslated digits. If the total number of digits exceeds 31, the excess digits will be truncated.

Each Pretranslation Table reduces the number of available Speed Call lists in the system.

Speed Call Controllers do not have access to Pretranslation Tables. Lists must be created and maintained through Service Change.

Before configuring a Pretranslation Data Block in LD 18, Pretranslation group 0 must be configured.

When Pretranslation is allowed in LD 15 (PREO = 1), in order for a pretranslation entry to be removed, Pretranslation must first be disabled in LD 15 (PREO = 0). The Pretranslation data block is then removed in LD 18. It is not possible to remove a single entry. The entire data block must be removed.

Feature interactions

Authorization Code Security Enhancement

The first digit dialed after a valid Authorization Code is sent to the pretranslator.

Automatic Redial

Automatic Redial (ARDL) can be activated on a number that has passed the Pretranslation process. However, on an ARDL call the Pretranslation process is not used.

Automatic Trunk Maintenance
Private Line
Telset Messaging

Pretranslation cannot be used with these features.

Automatic Wake Up

When the Pretranslation feature is equipped with AWU, the actual DN, not the pretranslation DN, should be used when programming the AWU call request.

Call Detail Recording

If a number dialed is pretranslated, the translated digits appear in the Call Detail Recording (CDR) records, not the dialed digits.

Call Forward

The DN dialed-forwarded calls are pretranslated.

Charge Account, Forced

The first digit dialed after a valid Charge Account Code is sent to the pretranslator.

Controlled Class of Service, Enhanced

The DN used to program the Controlled Class of Service (CCOS) should be the actual DN before pretranslation. When programming CCOS, the DN entered is not pretranslated.

Digit Display

The Pretranslation digit is displayed as it was dialed, but if the call is put on hold, the digits of the pretranslated DN are displayed.

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Pretranslation feature is supported in a DPNSS1 UDP network. At the originating node, the first digit dialed of a call is pretranslated to trigger the look-up of the stored code for the dialed digit in the pretranslation table associated with the calling terminal.

Direct Inward System Access

Direct Inward System Access calls are automatically assigned XLST 0.

Direct Private Network Access

Digits automatically inserted by Direct Private Network Access Digit Insertion are pretranslated during call processing in the same manner as if the caller had manually dialed the digits.

Electronic Switched Network

The pretranslator is used with calls to HNPA, HLOC, and Home CDP locations.

Flexible Feature Codes

Flexible Feature Codes must be accessible through a Pretranslation Table entry in order for users to activate features in this manner.

The Flexible Feature Code (FFC) feature will not be affected if the FFC's begin with "*" or "#", since before translation begins if the first digit is an "*" or "#" pretranslation will not be done. If any digits follow the FFC code, the first of the digits that follows will be pretranslated.

Forced Charge Account

The first digit dialed after a valid Charge Account Code is sent to the pretranslator.

Meridian Hospitality Voice Services

Prior to Meridian Hospitality Voice Services (MHVS), the setup of calls using the Applications Module Link (AML) was not supported from telephones using the Pretranslation feature. With MHVS equipped, call setup using the AML is supported.

Meridian Link Calls

Pretranslation cannot function with Meridian Link calls if the Hospitality Voice Services (HVS) package is enabled.

Special Prefix

The SPRE code must be accessible through a Pretranslation Table entry in order for users to activate features in this manner.

Speed Call
Speed Call, System

Entries must be accessible through a Pretranslation Table entry in order to place a speed call.

A Speed Call List number should be programmed to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

User Selectable Call Redirection

If Pretranslation (package 92) is enabled, the digits entered as the redirection DN are pretranslated before they are stored. Note that no Pretranslation occurs when the redirection DNs are used in such call processing features as Hunting or CFNA, eliminating the possibility that the redirection DN is pretranslated twice.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Allocate sufficient Speed Call lists to be used as Pretranslation Tables.

2 LD 18 – Add or change a Speed Call list to be used for each Pretranslation calling group.

3 LD 18 – Add or change the Pretranslation data block, defining the calling group to Speed Call list correlation. This list must be configured before Pretranslation (PREO) is enabled in LD 15.

4 LD 15 – Activate Pretranslation and define calling groups to Speed Call list correlation.

5 LD 10 – Associate an Analog (500/2500 type) telephone with a Pretranslation group.

6 LD 11 – Associate a Meridian 1 proprietary telephone with a Pretranslation group.

LD 17 – Allocate sufficient Speed Call lists to be used as Pretranslation Tables.

Prompt	Response	Description
REQ	CHG	Change.

TYPE	CFN PARM	Configuration Record. System Parameters
...		
- MSCL	(0)-8191	Maximum number of Speed Call lists.

LD 18 – Add or change a Speed Call list to be used for each Pretranslation calling group.

Prompt	Response	Description
REQ	NEW	Add, or change.
TYPE	SCL	Speed Call data block.
LSNO	0-8190	Number of Pretranslation list.
DNSZ	4-(16)-31	Number of digits that can be in a list entry.
SIZE	10	Maximum number of entries.
WRT	(YES) NO	Data is correct and can be updated in data store.
STOR		x is the first digit dialed.
	x *	* = block call.
	x ***	*** = delete the digit.
	x space <CR>	space <CR> = pass digit unchanged.
	x yyyy...y	yyyy...y = replacement digits.
WRT	(YES), NO	Data is correct and can be updated in data store.
STOR	<CR>	Ends input of list entries.

LD 18 – Add or change the Pretranslation data block, defining the calling group to Speed Call list correlation. This list must be configured before Pretranslation (PREO) is enabled in LD 15.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	PRE	Pretranslation

CUST	xx	Customer number, as defined in LD 15
XLAT	xxx yyyy	Pretranslation list, where: xxx = Pretranslation calling group number (0-254), and yyyy = corresponding Speed Call list number (1-8190).

LD 15 – Activate Pretranslation and define calling groups to Speed Call list correlation.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- PREO	0 1	Allow or deny Pretranslation, where: 0 = no Pretranslation, and 1 = Pretranslation.

Note: When Pretranslation group 0 is configured, care must be taken to define the XLST prompt, rather than letting it default automatically to 0. If XLST does default to 0 when Pretranslation group 0 is configured, all sets in the switch are affected.

LD 10 – Associate an Analog (500/2500 type) telephone with a Pretranslation group.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XLST	0-254	Associate telephone with the specified Pretranslation group.
	<CR>	Default to Pretranslation group 0 (only when REQ = NEW). It is important to define the XLST prompt, rather than letting it default to 0, as when Pretranslation group 0 is configured, all sets in the switch are affected.

LD 11 – Associate a Meridian 1 proprietary telephone with a Pretranslation group.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XLST	0-254	Associate telephone with the specified Pretranslation group.
	<CR>	Default to Pretranslation group 0 (only when REQ = NEW).

Feature operation

No specific operating procedures are required to use this feature.

Preventing Reciprocal Call Forward

Contents

This section contains information on the following topics:

Feature description 393
Operating parameters 394
Feature interactions 394
Feature packaging 394
Feature implementation 395
Feature operation 395

Feature description

This feature provides a modification to the Call Forward All Calls feature as a customer option. If set A attempts to enter a new Call Forward All Calls to set B, this modification verifies that set B has not been call forwarded to set A.

The verification process is repeated until one of the following conditions is met:

- the entered DN is not call-forwarded to any other set
- the activating set call forwards to the original Call Forward DN
- the maximum number of hunt steps is encountered a trunk is encountered, or
- a Pilot DN is encountered.

If a Multiple Appearance DN is encountered during the verification process, the only possible Call Forward Chain is checked.

Operating parameters

The verification is done only to current Call Forward states of the DNs being checked.

A set cannot Call Forward to itself.

This modification does not apply:

- to Hunt DNs
- to calls forwarded to the attendant
- across trunks

This feature applies to network environments.

Feature interactions

Network Call Redirection

For Network Call Redirection, when a call forwarding loop from one node to another occurs, the maximum number of redirections can be defined by the customer.

Remote Call Forward

This modification applies to Remote Call Forward.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Allow or deny Preventing Reciprocal Call Forward for a customer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options.
...		
OPT	(PVCA) PVCD	Enter PVCD to (allow) deny Preventing Reciprocal Call Forward.

Feature operation

If set A attempts to enter a new Call Forward All Calls to set B, verification is given that set B has not been call forwarded to set A.

When this situation is encountered:

- If the attempt to enter the new Call Forward DN was made on set A using a SPRE or Flexible Feature Code (typically on a 500/2500-type set), overflow tone is given to set A and the existing call-forward DN remains unchanged.
- If the attempt to enter the new Call Forward DN was made on set A using the Call Forward All Calls feature key, the attempted entry is treated like a normal invalid DN entry (that is, when the Call Forward All Calls key is pressed a second time after the DN has been entered, the associated lamp continues to flash until a valid forward DN is entered or the key is pressed for a third time).

Prime Directory Number

Contents

This section contains information on the following topics:

Feature description 397
Operating parameters 397
Feature interactions 398
Feature packaging 398
Feature implementation 398
Feature operation 398

Feature description

The bottom key on a Meridian 1 proprietary telephone is the Prime DN. It is preselected for call origination. If a user wishes to place or receive a call on any other DN, the key must be manually selected.

Operating parameters

Prime DN applies only to Meridian 1 proprietary telephones. Only one Prime DN is allowed per telephone.

Feature interactions

Automatic Wake Up FFC Delimiter

If you press the Prime Directory Number, when programming a Wake up request, you cancel the programming sequence. If an invalid timer is entered, the user hears an error tone. If another feature key is pressed during programming, it is ignored by the system.

Hot Line

If the Hot Line key is assigned to key 0 on a Meridian 1 proprietary telephone, it acts as the prime DN. When the user goes off-hook without selecting a DN key, the Hot Line is activated and the call is placed without further user action.

Feature packaging

This feature is included in base System Software.

Feature implementation

Assign key 0 as the Prime DN in LD 10.

Feature operation

No specific operating procedures are required to use this feature.

Privacy

Contents

This section contains information on the following topics:

Feature description 399
Operating parameters 399
Feature interactions 400
Feature packaging 401
Feature implementation 401
Feature operation 401

Feature description

Meridian 1 proprietary telephones automatically provide Privacy for telephones sharing a single call arrangement Directory Number (DN). When a call is in progress on the DN, no other telephone on which the DN appears can enter the call.

Operating parameters

Privacy is not available for analog (500/2500 type) telephones.

If the Directory Number (DN) is shared with any single line telephone, Privacy is not in effect for any appearance of the DN, and anyone sharing that DN can enter an active call.

Feature interactions

Automatic Redial (ARDL)

If the ARDL call is redialed on a number that is shared with any single line telephone, the ARDL call is accepted when the single line telephone goes off-hook.

Bridging

Privacy is lost when telephones are bridged. Any appearance of the DN can enter the call by going off-hook.

Call Hold, Permanent

A call placed on Permanent Hold has Privacy removed. Privacy is reinstated when the call is removed from Permanent Hold.

Group Call

The maximum number of DNs that can be added as members of a Group Call is 20. Each Multiple Appearance, Multiple Call Arrangement with Ringing (MCR) or Multiple Call Arrangement without Ringing (MCN) DN reduces the number of telephone sets that can be added to a Group Call. For example, if two telephones have the same MCR appearance of a DN, the number of telephones in the Group Call becomes 19. That is, each appearance of a DN counts as one member, up to a maximum of 20, of the Group Call.

> ***Note:*** Multiple Appearance, Single Call Arrangement with Ringing (SCR) or Single Call Arrangement without Ringing (SCN) DNs count as one member of a Group Call, irrespective of its number of DN appearances.

Multiple Appearance Directory Number

If a Multiple Appearance, SCR/SCN DN is shared by Meridian 1 proprietary telephones only, Privacy is in effect. No one can enter a call unless the call is first placed on Hold, or unless Privacy Release is activated to allow another appearance to enter the call. If this configuration is shared between these telephones and single-line telephones, Privacy is not in effect for any appearance of the DN. Anyone sharing the DN can enter the call at any time.

Privacy Override

The user can Override the inherent privacy on Meridian 1 proprietary telephones. If an appearance occurs on a telephone with Privacy Override enabled, that appearance can bridge into an active call. This pertains to calls on a multiple appearance single call Directory Number (DN) when not mixed with single line telephones.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Privacy Override

Contents

This section contains information on the following topics:

Feature description 403
Operating parameters 404
Feature interactions 404
Feature packaging 405
Feature implementation 405
Feature operation 405

Feature description

A Meridian 1 proprietary telephone with a Privacy Override Allowed (POA) Class of Service can enter an established call on a multiple appearance single call Directory Number (DN). However, the call cannot be joined until it is established (that is, the EOD timer has expired).

If all members of a non-mixed multiple appearance single call DN group are allowed Privacy Override, the operation of the feature is equivalent to a mixed multiple appearance single call arrangement.

When a group contains a combination of Privacy Override Allowed (POA) and Privacy Override Denied (POD) Classes of Service, the telephones denied Privacy Override cannot bridge into established calls.

Operating parameters

Privacy Override does not apply to analog (500/2500 type) telephones.

The system must be equipped with a conference loop. The number of timeslots is limited to 30 per conference loop. For Small Systems and Succession 1000 systems, a maximum of six parties per conference is supported.

Feature interactions

Automatic Redial (ARDL)

When the Privacy Override feature is activated on the MADN key and the one set activates ARDL, this call can be accepted by other sets.

Call Park
Call Transfer

Calls in a Privacy Override conference state cannot be parked or transferred.

Conference

The Conference feature can be used to add other parties to a Privacy Override connection.

Exclusive Hold

Telephones with POA Class of Service cannot bridge into calls on Directory Numbers (DNs) with Exclusive Hold active.

Multiple Appearance Directory Number - Mixed Mode

Since the Privacy feature is not active in this mode, telephones with a POD Class of Service can bridge into an active call.

Privacy

The user can Override the inherent privacy on Meridian 1 proprietary telephones. If an appearance occurs on a telephone with Privacy Override enabled, that appearance can bridge into an active call. This pertains to calls on a multiple appearance single call Directory Number (DN) when not mixed with single line telephones.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Allow or deny Privacy Override on a Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	POA (POD)	Allow or deny Privacy Override.

Feature operation

To activate Privacy Override, press the multiple appearance single call DN. You are automatically connected to the call.

Privacy Release

Contents

This section contains information on the following topics:

Feature description 407
Operating parameters 407
Feature interactions 408
Feature packaging 409
Feature implementation 409
Feature operation 409

Feature description

In multiple appearance single call arrangements of Meridian 1 proprietary telephones, Privacy Release allows one other appearance of the Directory Number (DN) to enter the call. Privacy is then reestablished until Privacy Release is activated again.

Operating parameters

Privacy Release is available only with Meridian 1 proprietary telephones in multiple appearance single call arrangements.

The system must be equipped with a conference loop. The number of timeslots is limited to 30 per conference loop. For Small Systems and Succession 1000 systems, a maximum of six parties per conference is supported.

Feature interactions

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Privacy Release (PRS) key is ignored if pressed.

Call Park

When a call from a Meridian 1 proprietary telephone is parked, that telephone cannot activate Privacy Release. For example, Party A calls Party B. Party B parks the call. Party A cannot activate Privacy Release.

China – Attendant Monitor

If Privacy Release is activated on a set that is involved in a monitored call, Attendant Monitor is deactivated.

Dial Access to Group Calls
Group Call

The Privacy Release feature cannot be applied to Dial Access to Group Calls and Group Call.

Exclusive Hold

If the telephone with Privacy Release has Exclusive Hold Allowed in the Class of Service, and a call is on hold, another telephone with that Multiple Appearance Directory Number (MADN) cannot access the call.

Multiple Appearance Directory Number

Privacy Release has no effect on Multiple Appearance, Multiple Call Arrangement with Ringing (MCR), or Multiple Call Arrangement without Ringing (MCN) calls.

Music, Enhanced

When using Privacy Release to add one or more members to a call already receiving Music, the Music is removed.

Ring and Hold Lamp Status

If the Privacy Release feature is activated for multiple-appearance single-call DNs, the blinking rate is based on the Class of Service of each set on which other appearances of the DN occur.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Allow/deny Privacy Release for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx PRS	Add a Privacy Release key. M2317 and M3000 telephones automatically assign the PRS key to key 28.

Feature operation

To allow someone with another appearance of the Directory Number (DN) to enter a call:

1. Press **Priv Rls**. All appearances of that DN flash. One other party can enter the call by pressing the flashing DN key that has the call.
2. You must press **Priv Rls** again to allow another appearance of the DN to enter the call.

Private Line Service

Contents

This section contains information on the following topics:

Feature description 411
Operating parameters 412
Feature interactions 413
Feature packaging 414
Feature implementation 414
Feature operation 417

Feature description

Private Line Service enables the customer to assign private Central Office (CO) lines to selected telephones or power fail transfer equipment. When associated with a Meridian 1 proprietary telephone, the following features are available to Private Line Service:

- Automatic Dialing
- Automatic Preselection
- Call Pickup
- Call Transfer
- Call Status
- Conference
- Common Audible Signaling

- Hold
- Multiple appearance single call arrangement
- Prime Directory Number
- Privacy
- Privacy Release
- Release, and
- Analog (500/2500 type) telephone/SL-1 telephone mix.

Operating parameters

Single line telephones with Private Line Service cannot access system features.

A maximum of 126 Private Lines are available per customer.

A Private Line should not be assigned as a Prime Directory Number (DN) unless preselection is required.

Hunting does not apply to Private Line service.

Call Forward on Private Lines (Meridian 1 proprietary telephones) is not forwarded to a second appearance of its own DN.

Feature interactions

Call Modification Features (CMF) in the trunk data block can be inhibited as follows:

- Call Transfer
- Conference
- Call Forward, and
- Message Center.
- Call Forward No Answer
  Call Forward No Answer is always inhibited on Private Lines.
- Multiple appearance
  For multiple appearance calls, call modification cannot be blocked.

Automatic Line Selection

A Private line DN is selected by Incoming Ringing/Non-Ringing Line Selection and Outgoing Line Selection.

Automatic Redial

An Automatic Redial (ARDL) call can be activated on a Private Line Service key. The call can only be redialed when the calling party's PVR or PVN key is free.

Call Park

Private lines cannot park a call.

Calling Party Privacy

The Private Line Service feature will outpulse the Privacy Indicator only if it is dialed by the originator. An asterisk will be outpulsed to the far end only if it is an Outpulsing of Asterisk and Octothorpe (OPAO) call; otherwise the asterisk signals a three-second pause.

China – Attendant Monitor

Attendant Monitor is blocked from monitoring a Private DN.

Collect Call Blocking

If an incoming DID or CO call from a private line trunk terminates on a set with a CCBA Class of Service, the Collect Call Blocking answer signal is provided in place of the regular answer signal.

Do Not Disturb

Do Not Disturb cannot be used on Private Lines.

Flexible Feature Code Boss Secretarial Filtering

Flexible Feature Code Boss Secretarial Filtering takes precedence over Private Line and Hot Line.

Hot Line

A Hot Line key cannot be a Private Line, as this would defeat the benefits of Private Line service.

Station-to-Station Calling

You must go over the public network to reach a Private Line. The software PRDN is not meant to be dialed directly.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 16 – Add or change a Private Line trunk route.
2. LD 14 – Add or change Private Line trunks in the Private Line trunk route.

3 LD 10 – Add or change Private Line Service for analog (500/2500 type) telephones.

4 LD 11 – Add or change Private Line Service for Meridian 1 proprietary telephones.

LD 16 – Add or change a Private Line trunk route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	COT	Central Office trunk.
PRIV	YES	Route is a Private Line route.
AUTO	(NO) YES	Trunks in this route autoterminate.
ICOG	IAO	Incoming and outgoing route.

LD 14 – Add or change Private Line trunks in the Private Line trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	COT	Central Office trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	XUT XEM	Universal Trunk Card (NT8D14), E&M Trunk Card (NT8D15). Prompted only for Superloops and the first unit on the card.

PRDN	xxx...x	Private Line phantom DN.
CMF	(NO) YES	Call modification is or is not inhibited for private line.

LD 10 – Add or change Private Line Service for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DN	xxx...x	Private Line DN (xxx...x is the same as for PRDN prompt in LD 14).

LD 11 – Add or change Private Line Service for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx PVN yyy...y	Private Line non-ringing key (yyy...y is the same as for PRDN prompt in LD 14).
	xx PVR yyy...y	Private Line ringing key (yyy...y is the same as for PRDN prompt in LD 14).

Feature operation

No specific operating procedures are required to use this feature.

Public Switched Data Service

Contents

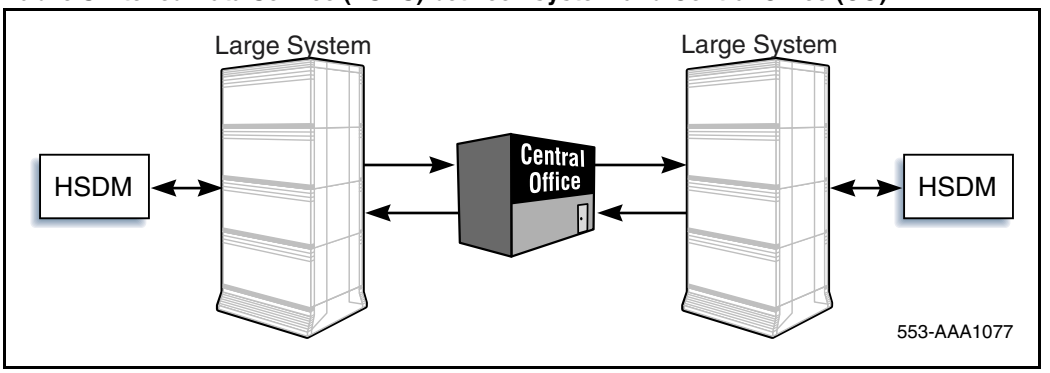
This section contains information on the following topics:

Feature description 419
Operating parameters 420
Feature interactions 421
Feature packaging 423
Feature implementation 423
Feature operation 423

Feature description

The Public Switched Data Service (PSDS) allows you to receive data on your system at 64 kbps over an Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) channel. See Figure 6.

Figure 6
Public Switched Data Service (PSDS) between system and Central Office (CO)



You can install a T1 link to different vendors and use the Meridian Communications Adapter (MCA) or QMT21 High Speed Data Module to initiate or receive a 56 kbps digital data call. The digital data call then transports across the vendor's digital network to another system or an SL-100.

Operating parameters

PSDS calls are supported in the following situations:

- a system and the Central Office (CO)
- a tandem call from an SL-100 to the system, and
- the system and other PSDS-compatible switches.

The PSDS supports Digital Trunk Interface (DTI) type trunks, TIE and DID/DOD trunks, and Electronic TIE Network (ETN) compatible signaling.

End-to-End DTI network

For all system networks (Point to Point), users can access the existing data facility in the system to support data calls, or they can select the Switched 56 data mode. For mixed-vendor private networks, users can only select the PSDS mode.

Feature interactions

ISDN PRI

The following routes are possible using this feature on Primary Rate Access:

- Point to Point access
  For Point to Point access of TIE trunks, the software can be modified to handle the requirements of this feature.
- Tandem call
  For tandem access, additional information on this feature is needed, or the data call can be defined as a voice call.
- DID/FEX/WATS/Accunet
  The system supports PSDS data calls to these trunk types.
- Public Network hop off
  Signaling is provided to inform the tandem switch about the PSDS data call.

Related features

When using PSDS, you may want to refer to the following features.

Meridian Communications Adapter (MCA)

The Meridian Communications Adapter (MCA) allows asynchronous ASCII terminals, personal computers, and printers to be connected to the telephone using an RS-232C or V.35 interface. The MCA also allows synchronous applications (DTEs such as video conferencing equipment and Group IV fax units) to be connected to the telephone. Refer to *Meridian Communications Unit and Meridian Communications Adapter: Description, Installation, Administration, Operation* (553-2731-109) for detailed information on the MCA.

Meridian Communications Unit (MCU)

The Meridian Communications Unit (MCU) provides a standalone version of the Meridian Communications Adapter (MCA).

The Meridian Communications Unit (MCU) allows you to transmit and receive data using either PSDS over the public network or a private network. The MCU, which replaces the QMT21C, is designed for domestic and international use, with transmission speeds up to 19.2 kbps asynchronous, and 64 kbps synchronous, integrated display, and self diagnostics. The MCU supports autodialing, ring again, and speed calling, as well as autobauding and automatic parity detection. You can use the MCU for:

- Video conferencing
- LAN bridging
- Bulk data/PC file transfer
- Dial back-up, and
- Host connectivity.

The MCU fully complies with RS-232C and can be configured as DCE or DTE to connect to a terminal, printer, or fax machine.

Unlike the MCA, the MCU provides a dedicated call key and call progress tones. The MCU also permits smart modem pooling.

The MCU supports the DM-DM, T-Link, V.25 bis, and PSDS interfaces as well as the RS-232C, CCITT V.35, CCITT V.24, and RS570/RS3449 (with different cables) interfaces. It complies with V.28 for European approval.

Refer to *Meridian Communications Unit and Meridian Communications Adapter: Description, Installation, Administration, Operation* (553-2731-109) for detailed information on this feature.

Transparent Data Networking (TDN)

Transparent Data Networking provides a transparent data channel for data modules to perform end-to-end protocol exchange. This means that two data modules will wait for a circuit path to be established before exchanging protocol parameters.

The data modules and protocols that are supported by TDN are:

- Meridian Communications Adapter (MCA) card in a Meridian Modular telephone (MMT) set, which uses PSDS and T-Link protocols on external calls

- Meridian Communications Unit (MCU), a standalone version of the MCA, which uses T-Link and PSDS protocols on external calls
- Basic Rate Interface (BRI) telephones, which use T-Link, V.110, and V.120 protocols, and
- High Speed Data Module (HSDM) when configured to use PSDS.

Refer to *Transparent Data Networking* (553-2731-110) for detailed information on TDN.

Feature packaging

This feature is included in base System Software.

Feature implementation

The data selection (DSEL) in the Route Data Block can be defined as voice calls only (VCE), data calls only (DTA), or voice and data calls (VOD). The call can be defined as voice calls, regular data calls, or PSDS calls. Refer to the *Software Input/Output: Administration* (553-3001-311) to configure the Route Data Block.

Feature operation

Originating data calls

For direct access, dial the regular seven-digit or 10-digit number.

For special route access, dial a route access code after hearing a dial tone.

Receiving data calls

Calls are answered automatically.

An auto-answer call is answered by the data module, and no special operation is necessary.

Pulsed E&M DTI2 Signaling

Contents

This section contains information on the following topics:

Feature description 425
Operating parameters 425
Feature interactions 426
Feature packaging 427
Feature implementation 427
Feature operation 432

Feature description

This feature provides pulsed channel associated ABCD-bit line signaling on 2 Mbps digital trunks. This signaling is used by the French Colisée and Indonesian systems, and is equivalent to analog pulsed E&M signaling. Pulsed E&M 2 Mbps Digital Trunk Interface (DTI2) signaling can be configured by using LDs 16 and 73.

Operating parameters

This feature does not apply to Small Systems and Succession 1000 systems.

Firmware changes to the QPC915C (French Colisée Pulsed E&M DTI2 signaling pack) and the QPC536E DTI (Indonesian Pulsed E&M DTI2 signaling pack), to implement the timing requirements of successive signals for both French Colisée and Indonesia.

Feature interactions

China Number 1 signaling

Cancel Offering (Toll Operator Break Out) is added to the Toll Operator Break-in feature. Calling Party Control is enhanced to use the OHTT, as well as the OHT prompt in LD 16.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Pulsed E&M is not supported by CIS DTI.

2 Mbps Digital Trunk Interface

Pulsed E&M DTI2 signaling is based on 2 Mbps DTI.

MFE for Socotel

Pulsed E&M DTI2 signaling is compatible with MFE for Socotel in the slave mode.

MFC/Semi-compelled MFC

Pulsed E&M DTI2 signaling is compatible with MFC and Semi-compelled MFC (SMFC).

New Toll Call Identification

Pulsed E&M DTI2 signaling is used to distinguish between national and international calls, in order to initiate clear back timing of the correct duration.

Periodic Pulse Metering

Pulsed E&M DTI2 signaling provides the following changes to PPM:

- the ANSWER and RE-ANSWER signals will be counted as a PPM pulse

- the counting of PPM pulses will not be activated when the call is set up; it will be activated when an ANSWER or RE-ANSWER signal is received, and
- PPM pulse detection will be turned off when a CLEAR BACK signal is received.

Lockout

Pulsed E&M DTI2 signaling will allow a flexible treatment to occur on outgoing trunks which are locked out. This will consist of allowing outgoing trunks which are locked out to send repeated FORWARD RELEASE signals.

Feature packaging

Pulsed E & M DTI2 Signaling requires the following packages:

- Pulsed E&M (PEDM) package 232
- International Supplementary Features (SUPP) package 131
- 2 Mbps Digital Trunk Interface (DTI2) package 129
- Special Services for 2500 Sets (SS25) package 18
- 500 Set Dial Access to Features (SS5) package 73
- Operator Call Back (China #1) (OPCB) package 126
- Attendant Break-in/Trunk Offer (BKI) package 127
- PPM/Message Registration (MR) package 101
- Multifrequency Compelled Signaling (MFC) package 128

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Configure the Route Data Block for Pulsed E&M DTI2 Signaling.

2 LD 73 – Configure the DTI Data Block for Pulsed E&M DTI2 Signaling.

3 LD 73 – Change the signal values for incoming/outgoing calls.

4 LD 73 – Change the signal values for incoming calls.

5 LD 73 – Change the signal values for outgoing calls.

LD 16 – Configure the Route Data Block for Pulsed E&M DTI2 Signaling.

Prompt	Response	Description
...		
RPPM	...	
A1MR		First Meter Pulse. Prompted if DTRK = YES, DGTP = DTI2 and MR = PPM.
	(NO) YES	Enter YES to cause the meter pulses received before an ANSWER signal to be invalid. The ANSWER signal is taken as the start of the first charging period (that is, when an ANSWER signal is received, the PPM count is incremented). NO is the default, and causes the meter pulses to be counted from the moment that the outgoing trunk is seized. When the trunk answers, the PPM count is left unchanged.
...		
IMCB	...	
TOBO		Toll Operator Break Out. Prompted if DTRK = YES, DGTP = DTI2 and MR = PPM.
	(NO) YES	If YES is entered, an OPCA signal received after a toll operator Break-in operation has been completed will result in the toll operator being removed off the call. If NO (the default) is entered, OPCA signals after a toll operator Break-in operation will be ignored.
...		
IHT	...	

OHT	0-(30)-62	Prompted if CNTL = YES and OPCB = YES. Enter the number of seconds, in increments of two, after which an outgoing CGPC non-toll call will disconnect, after the far end disconnects.
OHTT	0-(30)-62	Prompted if CNTL = YES and OPCB = YES. Enter the number of seconds, in increments of two, after which an outgoing CGPC toll call will disconnect, after the far end disconnects.
...		
FALT	...	
FRIN		Forward Release Indefinitely. Prompted only if DTRK = YES and DGTP = DTI2.
	(NO) YES	If YES is entered, a FORWARD RELEASE signal is re-sent every time the Disconnect Supervision timer expires and every time it is restarted. If NO (the default) is entered, a FORWARD RELEASE signal is not resent.
FRRC		Forward Release Repetition Count. Prompted only if FRIN = YES.
	0-(4)-15	Enter the value for the number of times that FORWARD RELEASE signal is resent before an error message is printed, if an acknowledgment is expected but not received.
FRRS		Forward Release Repetition Seize. Prompted only if FRIN = YES.
	(NO) YES	Enter YES to reseize the trunk before resending the FORWARD RELEASE signal. Enter NO to not have the trunk reseized before the FORWARD RELEASE signal is resent.
FRRD	128-(384)-1920	Forward Release Repetition Delay, in milliseconds. This is the delay between sending the SEIZE signal and FORWARD RELEASE signal. It is only prompted if FRIN = YES and FRRS = YES.

RRBS		Repeat Release Before Seize. This prompt allows a FORWARD RELEASE signal to be sent immediately before a SEIZE signal on a DTI2 trunk. Prompted only if DTRK = YES, DGTP = DTI2, and FRRS is not set to YES.
	(NO) YES	Enter YES to have a FORWARD RELEASE signal resent followed by the SEIZE signal. Enter NO to seize the trunk normally.
RLSM	(0)-15	Release Mechanism Only prompted if DTRK = YES and DGTP = DTI2.

LD 73 – Configure the DTI Data Block for Pulsed E&M DTI2 Signaling.

Prompt	Response	Description
...		
PERS	...	
DBNC	(10)-32	The De-bounce time for ABCD bit signals.
...		
TIME	...	
MINP	(8)-256	The Minimum Pulse Length for a Meter Pulse.
SASU	0-(1920)-32256	The Seize Acknowledge Supervision time, in milliseconds. ***Note:*** The JDMI default = 4992 milliseconds.

LD 73 – Change the signal values for incoming/outgoing calls.

Prompt	Response	Description
...		
FALT	...	
TIME	(0)-1920	The persistence time required before signal is accepted. ***Note:*** This value is used to implement the BLOCKING signal.

LD 73 – Change the signal values for incoming calls.

Prompt	Response	Description
...		
E SEZ(R)	ABCD	SEIZE signal.
TIME	16-(56)-1000 16-(296)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 56, and for off is 296.
E SEZA(S)	ABCD N	SEIZE ACKNOWLEDGE (answer) signal.
TIME	0-(150)-800	Delay, in milliseconds, before sending SEIZE ACKNOWLEDGE.
P WNKS(S)	ABCD N	Wink Start.
TIME	10-(220)-630	Pulse length of WNKS signal, in milliseconds.
P OPCA(R)	ABCD N	OPERATOR CALLING (receive) signal.
TIME	16-(96)-1000 16-(160)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 96, and for off is 160.
E CONN(S)	ABCD	CONNECT (answer) signal.
TIME	10-(150)-630	Pulse length of CONN signal, in milliseconds.
C CLRB(S)	ABCD/N	CLEAR BACK (answer) signal.
TIME	10-(600)-630	Pulse length of CLRB signal, in milliseconds.
P BRLS(S)	ABCD N	BACKWARD RELEASE (answer) signal.
TIME	10-(600)-2000	Pulse length of BACKWARD RELEASE signal, in milliseconds.
P FRLS(R)	ABCD N	FORWARD RELEASE (receive) signal.
TIME	16-(296)-2000 16-(960)-2000	Duration of pulsed time on and off, in milliseconds. The default for on is 296, and for off is 960.

LD 73 – Change the signal values for outgoing calls.

Prompt	Response	Description
...		
E SEZ(S)	ABCD	SEIZE signal.
TIME	10-(150)-630	Delay, in milliseconds, before sending SEIZE signal.
E SEZA(R)	ABCD N	SEIZE ACKNOWLEDGE (receive) signal.
P WNKS(R)	ABCD N	Wink Start (receive) signal.
TIME	16-(136)-504 16-(288)-504	Duration of pulsed time on and off, in milliseconds. The default for on is 136, and for off is 288.
E CONN(R)	ABCD	CONNECT (receive) signal.
TIME	16-(56)-1000 16-(296)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 56, and for off is 296.
C CLRB(R)	ABCD N	CLEAR BACK (receive) signal.
TIME	16-(296)-1000 16-(960)-1000	Duration of pulsed time on and off, in milliseconds. The default for on is 56, and for off is 296.
P FRLS(S)	ABCD N	FORWARD RELEASE (answer) signal.
TIME	10-(600)-2000	Duration of FORWARD RELEASE signal, in milliseconds.
P BRLS(R)	ABCD N	BACKWARD RELEASE (receive) signal.
TIME	16-(296)-2000 16-(960)-2000	Duration of pulsed time on and off, in milliseconds. The default for on is 296, and for off is 960.

Feature operation

No specific operating procedures are required to use this feature.

Radio Paging, X11

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Feature description

The Radio Paging (RPA) feature allows radio paging equipment (radio paging system) to be connected to a system. The radio paging system is a communications system used to contact mobile parties equipped with portable receivers. This communication is done via radio signals. The communication channels can be single-type (allowing one party to be paged at a time), or multiple-type (allowing several parties to be paged simultaneously).

To make a paging call, the calling party dials the paging access Flexible Feature Code. The paged party receives an indication of the incoming call in the form of a special tone, a verbal message, or a display message. The paged party can then answer the incoming call from any telephone set by dialing the answer paging Flexible Feature Code. The calling party remains off-hook until the call is answered. If all paging trunks are busy, the calling party receives a special congestion tone. The call can be tried again by redialing, or by activating the Ring Again feature.

When making a paging call, the system requires a paging access code, a mode digit, and dialed digit information. The paging access code is used by the paging system to identify the pager. The system derives this paging code by translating the DN of the party to be paged. This translation can be done in different ways, as described in this module. The mode digit indicates the type of display to be sent to the pager equipment (there are five possible display types). The digit information pertains to the calling party's DN. Depending on the type of paging chosen by the customer, this information is either entered manually by the calling party, or automatically by the system.

Local Radio Paging

To initiate a paging call, the Radio Paging System (RPS) requires the following activation sequence:

- Paging System Access (PSA) code
- mode digit
- information digits

The PSA code is the number used to identify a particular paging device. This code is derived by using the Directory Number (DN) of the party to be paged as a variable in the DN-PSA code translation procedure. If a valid DN is entered, the system sends the PSA code to the RPS that pages the party. If an invalid DN is entered, translation cannot be done and the caller receives Call To Vacant Number (CTVN) treatment. The caller can optionally page continuously until the following conditions are met:

- the paged party answers the page
- the caller goes on-hook
- the paging call times out

The paged party is required to answer the paging call within a specified time limit. When a paging call is not answered in time and the caller remains off-hook, a meet-me operation is possible. With this operation, calling parties to a radio pager are placed in a queue for a period of time, and the paged party can connect to the caller by dialing the answering Flexible Feature Code (FFC) and the paged party's DN. This connection appears as a simple call between two sets.

The paging time limits only apply to calls internal to the system. All external calls transferred to the RPA feature will be subject to the recall timer (not the normal attendant recall) if the call is not answered.

The paged party can answer a paging call from one of the following:

- A set connected to the system by dialing the answering FFC followed by their own DN in order to connect to the caller and free the paging trunk.
- A Public Switched Telephone Network (PSTN) telephone in order to contact the system attendant and request that the paging call be answered. The attendant dials the answering FFC followed by the DN to connect to the caller while the paged party is held on the Attendant Console's source-side. The two parties are then connected in the normal way.

When there are multiple paging calls to a pager, any attempt to page a party already engaged in a paging call will receive ringback (if configured) from the system or call progress tones from the RPS. The caller will continue to page until the paged party answers or the caller recalls.

Remote Radio Paging

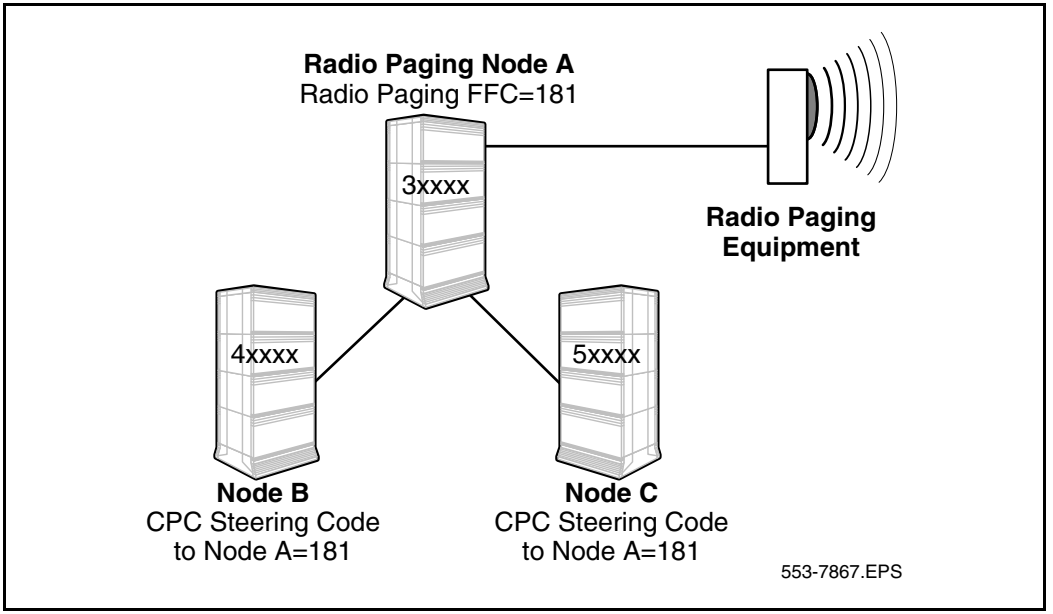
Remote Radio Paging (RRPA) provides a network-wide meet-me paging capability from a centralized location. Radio Paging can be accessed by remote nodes through a Coordinated Dialing Plan. These remote nodes can define CDP steering codes that route calls to the Radio Paging node.

Note 1: The Radio Paging (RPA) package is not required at remote nodes, unless post-selection Radio Paging is required.

Note 2: These steering codes are the equivalent of Flexible Feature Codes for Radio Paging, and are referred to as *Remote Radio Paging (RRPA) FFCs*. The steering codes must not be deleted by digit manipulation, since the digits are interpreted as the Radio Paging FFC at the radio paging node.

Figure 7 illustrates a possible Remote Radio Paging configuration:

Figure 7
A typical Remote Radio paging configuration



Node A, which is equipped with the Remote Radio Paging feature, is referred to as the Radio Paging node. The Radio Paging FFC is defined as 181. At remote nodes B and C, steering codes of 181 have been defined to route calls to node A. To access Radio Paging from nodes B and C, a caller simply has to dial 181.

Post Selection Access to Remote Radio Paging

Remote Radio Paging allows the *post selection* operation of Radio Paging from all nodes in the network. For this functionality, all nodes must be equipped with the Remote Radio Paging feature. For post-selection access, Trunk Steering Codes (TSCs) and Distant Steering Codes (DSCs) are defined as Remote Radio Paging (RRPA) FFCs.

If a post-selection access is made to a set on the same node, the originally-called set must be either ringing or busy. If the originally-dialed set is on another node, it must be on an established call. In this latter case, the established call is disconnected before being routed to the radio paging node.

Post-selection access can be performed from circuit switched network-type sets, Meridian 1000 series sets and Meridian proprietary sets, and Attendant Consoles.

Directory number to Paging System Access code translation

Each mobile paging device is identified by a unique Paging System Access (PSA) code. A single DN can only be translated to one PSA code. The following are the different types of translation methods available:

- no translation with DN sent as PSA code (single digits can be outpulsed immediately as dialed, or batched and sent all together)
- last two digits of DN sent as PSA code
- last three digits of DN sent as PSA code
- last four digits of DN sent as PSA code or
- a translation table is searched, and the stored PSA code for the DN is sent (several DNs can be associated with a single PSA code)

With the Group Hunting feature, it is possible to forward a call to a pilot DN which points to the table containing a list of DNs to be called. In this table the RPAC and DN for RPA can be stored.

Invalid directory number handling

With the first four methods, it is not possible for the system to detect if the DN is invalid. With the last method, an invalid DN is blocked with the caller receiving CTVN treatment. An individual with no telephone (or DN with which to associate) can use RPA through the use of a dummy DN. The method in which an RPS responds to an invalid PSA code varies by system.

Multiple Radio Paging systems

The RPA feature allows up to 16 (numbered 0 - 15) RPSs to be configured. The following are required to configure the RPA data block:

- The translation table is to be used for all systems.
- The DN is entered with respect to a particular system number.

Paging indications

The Radio Paging Access Code (RPAC), which is a defined FFC, allows access to the procedures required to initiate a paging call. After the access FFC is dialed, the caller receives the paging tone which is removed after the first digit of the DN is entered. Seizing the trunk to the RPS before or after dialing the DN, depends on the number of RPSs configured for the customer. After the initiating FFC and DN are entered, ringback can be provided or the RPS tones can be received.

If a trunk to the RPS is not available, the caller will receive the configured congestion busy tone. The call will have to be repeated when a trunk becomes available or the Ring Again feature is used (not for an inoperative RPS). The system will seize an idle paging trunk and send a PSA code to the RPS.

The following are cases where a tone from the system will be returned to the caller to indicate that paging is in progress:

- If ringback is not required, no tone is provided (some RPSs provide call progress tones to the caller).

- If ringback and detection-of-call-accepted signal are selected, then the caller gets the ringback tone (only after receiving a call accepted signal from the RPS).
- If ringback is required and detection-of-call-accepted signal is not required, then the caller gets the ringback tone after the valid entry of the FFC and DN.

When the caller is call forwarded (by CFNA or CFWAC) to a radio pager, a Recorded Announcement (RAN) can be sent to the caller.

Dialing plans

Two types of dialing plans can be used in a network:

- **Coordinated**. A single dialing plan is created to cover all the systems.
- **Independent**. Each switch has its own dialing plan, and the systems are connected by the use of RPACs.

The dialing plans can be arranged in various ways which can affect the way RPA works and how RPACs are manipulated.

With regard to dialing plans, the RPAC must be numeric to allow access from a second system. Also, the Calling Line Identification (CLID) is displayed if RPA is equipped, otherwise the route access code and member number are displayed.

Single paging system

This arrangement has two or more systems connected, but only one system (the source) has a RPS connected. Telephones connected to any connected system can page any party using the same RPAC. The paged party can answer a paging call from any telephone on the source system. A telephone on a non-source system can connect to a telephone on the source system by dialing the DN. If the call is redirected (for example, by ATT, CFW, or CFNA) the set on the non-source system can access the RPS.

Multiple paging systems

This arrangement has two or more systems connected, with each having a connected RPS. Different RPACs are required for each RPS (the user must be aware of which system is connected to which RPS). Trunk access between systems is handled by internal manipulation of the RPACs. When possible, RPSs should be connected to the same system.

Radio Paging system signals

The RPS has two categories of signals:

State of paging call

The following are the signals an RPS can send to the system in order to indicate the state of the paging call:

- A disconnect signal indicates that the paging trunk can be dropped.
- A call progress signal followed by a disconnect signal indicates a paging call is in progress.
- An all-digits-received signal indicates that all required digits are received.
- An absence signal, which is the receiving of a disconnect signal before a call progress signal, indicates that a pager is installed in the paging rack. (Calls to the pager in the rack receive the congestion tone from the system.).
- A paging-call-accepted signal indicates that the call is accepted.

Fault-clearing and maintenance

The system can interpret the ready-for-service signal from an RPS. The following system procedures occur when a fault on the paging hardware is detected:

1 All paging calls are dropped.

2 All trunks on the faulty system are made maintenance busy.

3 Subsequent paging calls on the faulty RPS will receive maintenance-busy treatment.

The following system procedures occur when the fault is corrected:

1 Idle all trunks on an RPS.

2 Each RPS is checked (faulty systems are made maintenance busy) after a system initializes and/or reloads.

Paging time limits

For sets internal to system network

Each RPAC has time limits defining how long a paged party has to answer a call. (The time limits only apply to sets internal to the system network, as external calls are subject to attendant (ATT) recall.) The following are the three paging timers:

- **Speechpath**. For the duration, the path is maintained.
- **Non-speechpath**. For the duration, a paging trunk is held to send digit information to the RPS.
- **Meet-me**. For the period of time to perform a meet-me operation, started after outpulsing is finished (interdigit timing is used for timing the DN entry).

The paging timers can be configured in the following two ways:

1 A warning tone is given eight seconds before a speechpath is dropped. After the speechpath timer expires, the trunk is dropped and the paged party is put under the meet-me timer. The caller is kept in a meet-me queue for this time.

2 The paging trunk is dropped if a paging-call-accepted signal is sent by the RPS. If a paging-call-accepted signal is not sent, after the non-speechpath timer expires the paging trunk is dropped. A meet-me timer then comes into effect.

   If a paging call is not answered before the meet-me timer is activated, the paging trunk is dropped (available for other calls) and the paging device stops paging.

   If a paging call is not answered after the meet-me timer has expired, the paging set is subjected to line lock-out procedures and ringback (if configured) to the caller is stopped.

For sets external to system network

The recall timer overrides the existing Attendant Recall on all external calls transferred to the paging trunks. The recall timer is required because a paged party is expected to take a longer time to answer a call. Any recall to the attendant is presented to the attendant as a recall Incoming Call Indicator (ICI). Forwarded calls to the RPS will recall to the attendant. External calls are transferred to the paging equipment by the following:

- **Attendant.** Defined in the RTSA feature.
- **Set**. For calls transferred by circuit switched network or SL-1 sets.

Methods of operation

Two different operational methods, automatic and manual, are available for RPA. Various RPACs are provided for in each method. Each RPAC has different options associated with it.

Automatic

The system sends all necessary digit information automatically for the caller. The digit information cannot be modified.

The following are the procedures for an RPA call:

1 Enter the RPAC.

2 Enter the DN of the paged party.

The system then transmits the following digit information to the RPS:

a. PSA code of the receiving device,

b. mode digit, and

c. the DN of the caller, if required (DN key used to page call).

Manual

The caller is required to enter the mode of operation that is desired. The caller sends any required digit information from the set.

The following are the procedures for an RPA call:

1 Enter the RPAC.

2 Enter the DN of the paged party (optionally translated to a PSA code).

3 Enter the mode digit.

4 Enter the necessary digit information.

5 Enter **#** to indicate the end-of-digit information.

The system then transmits the following digit information to the RPS:

a. PSA code of the receiving device

b. mode digit, and

c. all entered digit information.

Parallel paging

Parallel paging is a type of operation that applies to some TIE trunk interfaces (primarily used in Switzerland).

Parallel paging has the following characteristics:

- The caller remains off-hook until the paged party answers or until the call is terminated.
- The caller does not get any call progress tones from the RPS, only ringback from the system.
- The paged party's receiving device only has the capability of indicating that there is a call.
- Only the display bleep mode of operation is allowed.
- The caller receives no indication that a PSA code is invalid. The system supplies ringback tone until the call times out.

Initiating a paging call

Each of the following two procedures for initiating a paging call use the same RPAC, but require that the DN be dialed at different times.

Pre-selection

Radio Paging is accessed immediately by entering the RPAC and the DN. The caller knows the RPA feature is required before going off-hook.

Post-selection

The caller dials the DN before knowing that RPA is required. While receiving ringback or busy tone, the caller dials an RPAC (an FFC) to make the destination set stop ringing (the DN of the paged party does not have to be entered a second time).

When the caller puts a call on hold (For example, by Call Transfer or Conference key) and dials another set, post-selecting on Call Transfer or Conference is not allowed.

The automatic and manual methods of operation allow post-selection access to RPA. Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

The following are ways to perform post-selection access to RPA:

From a circuit switched network set

The caller sends a recall signal and receives a special dial tone, then dials the required RPAC or has single-digit access using the 16-digit post-selection feature. The caller receives Call to Vacant Number (CTVN) treatment if the RPAC is invalid.

From a Meridian 1 proprietary set

The caller presses the RPAG key (that has an RPAC associated with it) or 0 - 9 using single-digit post-selection to access RPA. The caller receives CTVN treatment if the RPAC is invalid.

From an Attendant Console

The caller presses the RPAG key (configured with an RPAC) to contact the paged party. The attendant receives no special dial tone, and the PAG key lamp is not used. When the RPAG key is pressed, the flashing SRC or DEST lamp becomes lit if the post-selection was successful, otherwise it remains flashing.

Modes of operation

A variety of modes, defined in mode digits, are available to allow the caller to send different types of digit information to the pager before completing the paging procedure. Some mode digits require additional information from the caller. The mode digits conform to the European Selective Paging Manufacturers Association (ESPA) standards. The caller can optionally receive call progress tones from the RPS while off-hook.

When the attendant extends a call to a pager that is in the rack, an absence signal is returned and the call is relinked into the attendant queue. When a telephone extends a call to a pager that is in the rack, the call is recalled to the set.

The following are the five mode digits:

Mode 1: External meet-me display

With Mode 1, the paged party receives a bleep and/or EXT is displayed (for external caller) on the pager. The external number or trunk route and member number are not sent by the system. The paged party accesses a telephone and enters the answering RPAC (an FFC) followed by their DN. The system connects the two parties.

Mode 2: Internal meet-me display

With Mode 2, the paged party receives a bleep and/or the caller's DN (1 to 7 digits) is displayed in the form MM*dn* on the pager. The paged party accesses a telephone and enters the answering RPAC (an FFC) followed by their DN. The system connects the two parties. Network (ISDN) calls are considered internal and display the set's Calling Line Identification (CLID).

Mode 3: Display bleep

With Mode 3, the paged party receives a bleep and/or the caller's DN (1 to 7 digits) is displayed in the form C*dn* on the pager. The paged party makes a simple call to the caller.

Mode 4: Two-way speechpath

With mode 4, the paged party receives a bleep and the caller's DN (1 to 7 digits) is displayed on the pager. A two-way speechpath (between the caller and pager) is created for a specified period of time.

Mode 5: Alarm display

With Mode 5, the paged party receives a bleep frequency and/or unique text is displayed (explaining the urgency of the call) and/or the caller's DN. The paged party makes a call to the caller.

Note: This mode is for emergency use only.

Terminating a paging call

The Radio Paging trunk can be released in the following four ways:

- The paged party answers the paging call by dialing the answering RPAC followed by their DN.
- The caller goes on-hook.
- The paging call times out.
- A disconnect signal is sent from the RPS.

Operating parameters

A maximum of 16 RPSs are allowed per customer.

The number of channels to the RPS is limited to the number of trunk members allowed for a trunk route.

A PSA code must be a minimum of one digit to a maximum of seven digits in length.

Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

Post-selection access at RRPA nodes is not supported on the ABCD keys of ABCD sets.

All DNs in the network must have the same fixed length.

The RPA feature is offered to each system disk as a package only.

The translation table size is restricted by the amount of memory available.

The serial type of paging is not supported.

The RPA feature is not available within a Dial Intercom Group (DIG).

The Multi-party Operations (MPO) Three-party Service does not work while RPA is in progress.

Call transferring an RPA call to another party is not supported.

Adding an RPA call to a conference is not supported.

Since ISDN BRI telephones do not support FFCs, they cannot be used to access or answer RPA calls if the BRI telephones are local on the paging node. For network situations, BRI telephones can access and answer remote RPA calls. This is possible because the RPAX/RPAN FFCs are dialed as DSC/TSC steering codes.

For network RPA recall, the originating, tandem and paging nodes must be system switches.

For the Pre-selection to Paging situation, if the paged DN following the RPAX FFC is not local to the paging node, the CPND name for this DN cannot be obtained to display on the calling party. If the paged DN is local on the paging node and has CPND defined, the CPND can be retrieved and sent to the calling party for display purposes. For Post-selection to Paging, the CPND of the paged DN will be displayed even if the DN is not local to the paging node.

There is an existing option that allows the replacement of the RPAX FFC with a character string on set's displays. This is controlled by the DCHR prompt in LD 58. This only applies to the local paging node. On the remote node, the RPAX FFC is treated as DSC/TSC and therefore will be displayed as it is. This is an existing limitation of network Radio Paging and remains unchanged.

If a network call comes in to a set on the paging node and is redirected to paging by CFNA, the calling name cannot be retrieved and updated on the answering set when the paging call is answered. This happens only if the set on the paging node has CPND defined. If the set does not have CPND defined, the calling name could be updated on the answering party. This is a design limitation.

The following hardware is required for RPA operation:

- Televerket (TVT) Tateco system T-800 or T-900
- Hasler system DS-1000 or DS-2000
- trunk cards for parallel paging QPC*xxx* (TIE)

Feature interactions

Access restrictions

The RPA feature uses a TIE or Central Office (CO)/Public Exchange route to connect the system with the RPS equipment. This has some impact on current restrictions when the route is used for this purpose.

Class of Service restrictions

All restrictions that currently apply to TIE or CO routes do not apply if the route is used for Radio Paging. Any restricted set is capable of initiating an RPA call, while any set can be used to answer a paging call. The restricted set is capable of answering a paging call, even if it is from the exchange network.

Trunk Group Access Restrictions codes

The TIE or CO routes that are used for the RPA feature are subject to the limitations applied by Trunk Group Access Restrictions (TGAR) codes. Sets can be prevented from using RPA, but only after the RPAC entry. The restriction applies when accessing RPA and not when answering a call.

Trunk Barring

The normal trunk-to-trunk restrictions apply to the TIE or CO routes that are used for Radio Paging.

Attendant Recall

An RPA caller using a circuit switched network set cannot recall the attendant by flashing, as it is ignored.

An RPA caller using an SL-1 set cannot activate the ATT recall key, as it is ignored.

The Radio Paging (RPA) recalls to the local attendant on the node where the RPA system is directly connected. This product improvement enables RPA to recall the attendant who originated the Radio Paging call only; the attendant may be located anywhere within a Meridian Customer Defined Network (MCDN).

The improvement also allows the attendant's display to be updated with paged name and to display paged name instead of answering name on the paged party when answered. In addition, the improvement enables network Radio Paging to show the same display information as in standalone operation.

Automatic Call Distribution

An Automatic Call Distribution (ACD) agent is allowed to transfer a call to RPA. The following are the operations:

- When a recall takes place and the transferring party is an ACD agent, the call is recalled to the ACD queue.
- When an RPA call is answered before the recall is presented to an ACD agent, the recall is removed from the queue.
- When an RPA call is answered while recall is presented to an ACD agent, the ringing is removed and the ACD agent is idled for other calls.
- When an RPA call is dropped while recall is presented to an ACD agent, it appears to the ACD agent as if the call was answered.
- When an ACD agent with an RPA recall presented presses a DN or a Make Set Busy key, the recall is removed from that ACD agent and a new recall to the ACD agent is attempted. If no ACD agents exists, the call is recalled to the attendant.

  Note: It is not possible to answer an RPA call that has recalled to an ACD agent with the Call Force option.

Automatic Dialing

The Autodial key can be programmed to perform RPA.

Automatic Timed Reminders

A new RPA recall timer (longer duration) overrides the existing recall timer. This RPA recall timer applies only to Public Switched Telephone Network (PSTN) and direct inward dialing (DID) sets using RPA trunks. The call receives Recall To Same Attendant (RTSA) treatment if the paging call is not answered by the paged party within the specified time.

Barge-in

Barge-in to either a caller trunk or an RPA trunk, while RPA is in operation, is not permitted and results in an overflow tone being returned to the attendant. The RPA operation is not affected and the paging will continue until one of the following occurs:

- the caller goes on-hook;
- the call is answered; or
- the call times out.

If an attendant attempts to Barge-in to an RPA trunk that is not busy, the trunk is seized and a dial tone is returned to the attendant. The attendant can then dial a PSA code to page the desired party. The method of operation is the same as Barge-in to an idle trunk.

Basic Automatic Route Selection

Radio Paging CO and TIE trunk routes can be set up with BARS.

> ***Note:*** These routes should not be entered in a schedule with normal CO or TIE routes, because they will respond differently.

Break-in

Break-in to either a caller or paged party, while RPA is in operation, is not permitted and results in an overflow tone being returned to the attendant. The RPA operation is not affected, and paging continues until one of the following occurs:

- the caller goes on-hook;
- the call is answered; or
- the call times out.

Busy Verify

Busy Verify for either a caller or paged party, while RPA is in operation, is not permitted and results in an overflow tone being returned to the attendant. The RPA operation is not affected and the paging will continue until one of the following occurs:

- the caller goes on-hook;

- the call is answered; or
- the call times out.

Call Detail Recording

Call Detail Recording (CDR) has two types of operation:

CDR on incoming or outgoing calls to Radio Paging system

In the first type, no CDR S record (between trunk and transferred party) is printed until the call is answered. Upon disconnection of an answered paging call, a CDR E record (between trunk and paged party) is printed, identifying the paged party DN and not the DN of the set from which the call was answered. Call Detail Recording (CDR) for internal calls is consistent with CDR for external calls.

No CDR record is printed on paging recalls which are re-extended to the paging trunk.

CDR on paging route

An “S” record is printed when an attendant extends an outgoing trunk call to a destination party. When the extended outgoing trunk call or the destination party releases to disconnect, an “E” record is printed.

Call Forward
Call Forward All Calls

This feature can allow equipped circuit switched network or SL-1 sets to have calls automatically forwarded to an RPAC. This forwarded number can be numeric or a non-numeric version in the FFC table.

Forwarding internal and external calls to the RPS requires the call forwarding number be defined as the RPAC and DN of paging device. If just the RPAC is entered, the paging DN is that of the set where CFW is activated. The RPS can provide a RAN for the caller.

Call Forward No Answer

A call to a circuit switched network or Meridian 1 proprietary set that is not answered after a specific number of rings is automatically forwarded to an RPS.

Call Transfer

A call can be transferred to an RPS with the following conditions: internal calls are subject to paging time outs; and external calls are subject to recall.

When transferring a call to an RPS, the transferring party may use pre-selection or post-selection method of access.

Call transferring an RPA call to another party is not supported.

Central Office/Public Exchange trunks

Central Office/Public Exchange trunks can be used for transfer of information to an RPS when the call progress tones from the RPS are received.

Conference

While in a conference, a party can make a paging call by using one of the following: switchhook flash (from a Meridian 1 proprietary set), Transfer (TRN) key, or Conference (A06) key (from a BCS set).

When the RPA call is complete, the party can drop Radio Paging and return to the conference. Adding an RPA call to a conference is not supported.

Dial 1

Using the register recall on a circuit switched network set, while receiving ringback tone, is allowed. If register recall is not allowed for a user, a ground button is used to allow post-selection initiation.

Digit Display

SL-1 set

During RPA operation, the display shows the FFC and DN for pre-selection and the DN FFC for post-selection initiation. When a call is re-routed (forwarded, hunted or transferred) to the RPS, the caller's display shows the FFC and paged party DN. After a paging call is answered, the caller's display is updated to show the answering set’s DN. The paged party's set displays the caller's DN.

Attendant Console

The display is similar to the Meridian 1 proprietary set when accessing and answering RPA calls. When a recall from paging occurs, the Attendant Console display shows the RPA FFC and the paged party's DN. The recall ICI key also indicates that the paging call has recalled.

The CLID is displayed if that feature is equipped. With CPND, the paged party's name supplements their DN display. If the Display Characters (DCHR) option is used in the RPA (LD 58), the FFC DN is replaced by the specified characters.

Direct Inward Dialing

When an incoming DID trunk attempts to gain access to a TIE or CO trunk that is configured as having RPS equipment, these calls are not intercepted by the attendant. The RPA call is made in the normal manner. The RPAC must be numeric.

Direct Inward System Access

Public Switched Telephone Network (PSTN) calls, accessing the RPA trunk, are handled in the same fashion as direct inward dialing calls.

Do Not Disturb

A set (DN) in the Do Not Disturb (DND) state can receive paging calls.

Enhanced Flexible Hotline

The RPAC (FFC) and DN can be stored in a hotline list of pre-set digits.

Group Hunting

With Group Hunting, it is possible to forward a call to a pilot DN that points to a table containing a list of DNs to be called. In this Group Hunting table, the RPAC (FFC) and DN for RPA can be stored.

Hold

The Hold key or autohold works on a paging call as if a station-to-station call is being made. The caller's set can be on hold while receiving a ringback tone or call progress tones. When a paging call is put on hold, no indication is given if the call has been answered. The Attendant Console SRC lamp is continuously lit, from the winking state, when the call that is put on hold is answered.

Last Number Redial

When a valid RPA FFC with a DN is entered and the configured length is enough, the FFC and DN are stored. When a manual RPA FFC is entered, the information digits and octothorpe (#) character are also stored.

Multifrequency Compelled Signaling (MFC)

Radio Paging can be accessed by a diversion from TIE or DID trunks using MFC.

The idle signal is not sent immediately when the RPA trunk is seized, since the RPS answers with a call accepted signal or a busy signal (when the ACPS prompt is set to YES). An idle signal is sent back immediately when one of the following occurs:

- no signal can be sent back from the RPS (when the ACPS prompt is set to NO);
- a Recorded Announcement (RAN) is provided; or
- Recall on Busy is configured.

Multiple Appearance Directory Number

With a Multiple Appearance DN, only one receiving device PSA code can be associated with the DN (not associated with a particular set).

Multiple Customer Operation

Each customer can connect to the RPS equipment. The RPSs connected are independent of each other.

Multi-party Operations (MPO)

It is possible to hold an existing call (during Call Join, Three-party Service or Conference-6) and initiate or answer a paging call. Transferring an external call is subject to the RPA Recall timer. When there is no answer to an initiated paging call, the call is released in the normal manner by pressing the DN key again on a Meridian 1 proprietary set or pressing Register Recall on a circuit switched network set. The MPO user can toggle between an established call and a paging call.

> ***Note:*** Three-party Service does not work while RPA is in progress. If the caller flashes with an established held call and an active unanswered paging call, the paging call is stopped and the held call is reestablished as active.

Network Automatic Route Selection (NARS)

Radio Paging CO and TIE trunk routes can be set up with NARS.

> ***Note:*** These routes should not be entered in a schedule with normal CO or TIE routes because they will respond differently.

Night Service

Incoming calls to a Night Service set (DN) can be transferred to RPA DNs. Calls can be entered or answered from the Night Service set. External calls transferred to RPA DNs recall to the Night Service DN.

Override

This feature allows a set to break into an existing call. The Break-in feature restrictions apply.

Ring Again

The RPA feature allows Ring Again to be applied when a paging route is busy. The caller can re-apply Ring Again when the congestion tone is received.

With RPA post-selection access and a caller attempting Ring Again, the indications that Ring Again is already activated or the queue is too large cannot be given until the RPAC has been dialed.

With RPA pre-selection access to a single RPS, the busy trunk indication is given immediately after the RPAC (FFC) is dialed. Ring Again only redials the trunk (on Meridian 1 proprietary sets all digits entered after the busy tone are redialed). The DN to be paged has to be re-entered.

With RPA pre-selection access to multiple RPSs and RPA post-selection access to a single RPS or multiple RPSs, the busy trunk indication is given after the DN is entered. Ring Again redials the trunk and the DN (all digit information in the automatic method is also redialed). Ring Again is ignored when a set is forwarded to the RPS, and all the trunks are busy.

Slow Answer Recall

A paging call is recalled to the attendant if it has gone unanswered after a period of time. The attendant uses the RLS key to extend the call again. The attendant console displays the RPAC (FFC), DN and CLID when there is a recall from paging.

Slow Answer Recall Modification (SLAM)

With the Slow Answer Recall Modification feature enabled, when the attendant answers a recall, the destination party is disconnected.

When the attendant answers a paging recall, the call is removed from the meet-me queue and the recall cannot be answered by the paging party by using RPA Answer. The paging party is put on the source side of the attendant; there is nothing connected on the destination side. The attendant cannot extend the call to paging by pressing the Release key. Pressing the Release key will disconnect the paging party from the source side and the attendant will become idle.

The attendant can extend the call to Radio Paging again by either: dialing the RPAX FFC + the DN (preselection); or dialing the DN, and while the DN is ringing or busy pressing the RPAG key (post-selection).

Speed Call

The Speed Call feature can be set up to perform RPA dialing.

Station-to-station calling

When a party is paged by one caller and a second party dials the paged party's DN, the call will ring the paged party's set in the normal manner.

Switchhook Flash

Using the register recall on a circuit switched network set is allowed while receiving a Ringback tone. If register recall is not allowed for a user, a ground (earth) button is used to allow the post-selection access method.

Tenant Service

A tenant can be restricted from accessing an RPA trunk and can be configured to share or privately use an RPA trunk. All other restrictions apply to RPA.

TIE trunks

This trunk type is used for information transfer to an RPS. Special hardware is required.

Traffic Measurements

The following traffic measurements are available for RPA:

- **Paging recall count.** Incremented each time a paging call is recalled to the attendant.
- **Average answer time.** The average time paging calls are in the paging queue before being answered.

Trunk Group Busy Indication

The Attendant Console's Trunk Group Busy (TGB) key/lamp pair can be assigned to each of the RPA trunk routes. The attendant presses the TGB key to deny a set access to a RPS. The TGB lamp goes on and all calls to the RPS are routed automatically to the attendant. Normal RPS access returns and the lamp goes off when the attendant presses the TGB key again. The following conditions apply to sets with TGAR:

- Sets with TGAR of 0 to 7 are routed to the attendant if the trunk group being accessed has been made busy by the attendant.
- Sets with TGAR of 8 to 15 are not restricted by the TGB operation by the attendant.

The TGB lamp flashes when all trunks in the paging trunk group are busy.

When a RPS is faulty, its TGB lamp flashes after all associated (with the faulty paging route) trunks have been made maintenance-busy. The reverse happens when the fault is corrected in the RPS hardware.

Feature packaging

The following feature packages are required for paging operation in addition to the Radio Paging (RPA) package 187:

- Flexible Feature Codes (FFC) package 139 (to gain access to RPA);
- 16-Button Dual-tone Multifrequency Telephone (ABCD) package 144 (to allow single digit post-selection access to RPA);
- For the Radio Paging network recall operation, Network Attendant Service (NAS) package 159 must be provisioned;
- For Remote Radio Paging, Coordinated Dialing Plan (CDP) package 59 is required to define RPA FFCs as Distant Steering Codes (DSCs) or Trunk Steering Codes (TSCs);
- To display characters instead of the Radio Paging Flexible Feature Code, Calling Party Name Display (CPND) package 95 is required; and
- Integrated Services Digital Network (ISDN) package 145 and its dependencies are required for operation in a Meridian Customer Defined (MCDN) ISDN network.

Feature implementation

Adding a Radio Paging System

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable or disable the RPA feature.

2 LD 16 – Configure trunk route for Radio Paging feature.

3 LD 14 – Enable or disable the reversing of the E-lead.

4 LD 11 – Configure the RPAG key for SL-1 sets.

5 LD 12 – Configure the RPAG key for Attendant Consoles.

6 LD 56 – Configure the RPA warning tone.

7 LD 57 – Define the Flexible Feature Codes (RPACs).

8 LD 58 – Define RPA customer information.

9 LD 58 – Define RPS information.

10 LD 58 – Define the RPAC information.

11 LD 58 – Change the Translation Table Information.

12 LD 18 – Define the ABCD table.

13 LD 18 – Define the pre-translation and post-translation list numbers.

LD 15 – Enable or disable the RPA feature.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Features and options data block.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
RPA	(NO) YES	Radio Paging Allowed.

LD 16 – Configure trunk route for Radio Paging feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems

TKTP	TIE COT	Trunk route.
RPA	(NO) YES	Radio Paging Route.
OPR	(YES) NO	Outpulsing Route (YES is the default if RPA = YES).

LD 14 – Enable or disable the reversing of the E-lead.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	TIE COT	TIE trunk. Central Office trunk.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
CLS	RVEP XREP	Reverse earpiece. Do not reverse earpiece.

LD 11 – Configure the RPAG key for SL-1 sets.

Prompt	Response	Description
REQ:	CHG	Change RPAG key assignment.
TYPE:	aaaa	Telephone type, where xxxx = SL1, 2006, 2008, 2009, 2016, 2112, 2216, 2317, 2616 or 3000
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx RPAG yyyy	To define an RPAG key with an RPAC (FFC), where xx is a key number and yyyy is an RPAC.

LD 12 – Configure the RPAG key for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	xxxx	Type of Attendant Console. xxxx = 1250, 2250.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx RPAG yyyy	To define an RPAG key with an RPAC (FFC), where xx is the key number and yyyy is an RPAC.

LD 56 – Configure the RPA warning tone.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	FTC	Flexible Tone and Ringing data block.
TABL	0-31	FTC Table Number.
SCCT	(NO) YES	Modify Software Controlled Cadences and Tones.
RPAW	x xx xx xx	Radio Paging Warning tone definition.

LD 57 – Define the Flexible Feature Codes (RPACs).

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	FFC	Flexible Feature Codes Data Block.
CUST	xx	Customer number, as defined in LD 15
CODE	RPAX	Radio Paging Access Code.

-RPAX	RPAX xxxx	Radio Paging Access Code. Enter Flexible Feature Code. The RPACs entered here are associated with various options in LD 58.
CODE	RPAN	Radio Paging Answer call code.
-RPAN	RPAN xxxx	Radio Paging Answer call code. Enter Flexible Feature Code.

LD 58 – Define RPA customer information.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RPCD	Radio Paging Customer Data Block.
CUST	xx	Customer number, as defined in LD 15
RPTO		Radio Paging Tone.
	SPCL DIAL NONE	Special Dialtone. Normal Dial tone. No Tone. This Radio Paging tone is provided after the RPAX and RPAN.
MRPS	(NO) YES	Multiple Radio Paging Systems.
TRAN		Translation type.
	TAB TWO THR FOR NO	Table Search. Last two digits of DN. Last three digits of DN. Last four digits of DN. None. Prompt is not given when MRPS = YES and TRAN is forced to TAB.
DNLN	1-(4)-7	DN length (if TRAN = NO, TWO, THR or FOR).

RCRG	0-(6)-20 X	Number of ring cycles when recall to transferring set, before reroute to attendant. (0 is the CFNA prompt value.) Reroute to attendant.
RCTI	0-(30)-120	Time to wait for a “BUSY” transferring set to become idle.
RCAL	(NO) YES	Recall if busy from RPA.
TBTR	4-(10)-30	Time between two recall attempts (to an SL-1 set).

LD 58 – Define RPS information.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RPS	Radio Paging System Data Block.
CUST	xx	Customer number, as defined in LD 15
SNUM	0-15	System Number.
PSAL	1-7	Paging System Access code length.
RTIM	0-(60)-630	Length of the Recall Timer.
STO	10-(30)-630	Length of time for Screech Path to be maintained in seconds.
NSTO	10-(30)-630	Length of time required for paging when no Screech Path is required.
MTO	0-(150)-630	Length of the Meet-Me Time-out timer in seconds.

LD 58 – Define the RPAC information.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RPAX	RPAC Data Block.
CUST	xx	Customer number, as defined in LD 15

SNUM	0-15	System Number.
RPAX	nnnn	Radio Paging Access Code.
-ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
-PANN	(NO) YES	Record Paging Announcement.
--RPAR	 0-511 0-127	Route Number that provides the recorded announcement. For Large Systems For Small Systems and Succession 1000 systems
-BYPS	(NO) YES	Bypass the DN-PSA translation. If BYPS = YES, then meet-me is not available, and the trunk is accessed directly.
--OPER	(AUTO) MANU	Automatic Operation. Manual Operation.
--EXTM	(0)-9 (If OPER = AUTO)	
--INTM	(0)-1-9	Internal Mode digit for this RPAX.
--TRDN	(0)-7 (If OPER = YES)	Transmit this number of digits of the caller's DN to the paging equipment.
-PATH	NONE SPCH RNGB	Speech Path or Ringback Speech Path. Ringback to the caller.
--TWSP	If PATH = SPCH (BOTH) EXT	Two-way Screech Path with a mobile pager allowed. Internal and external calls. External calls.
--ACPS	If PATH = SPCH (YES) NO	Radio Paging System to provide the call-in-progress signals.

--ACPT	If PATH = SPCH or RNGB, (YES) NO	Call Accepted is to be detected. When PATH = RNGB or SPCH, and ACPT = YES, Ringback is provided only when the call-accepted signal is received. Speech Path opens when the start-talk signal is received. When PATH = RNGB and ACPT = NO, Ringback is provided when all the paging digit information has been sent (ending # processed). When PATH = SPCH and ACPT = NO, Speech Path is provided when all of the paging digit information has been sent (ending # processed).
--DCHR	xxxx X	Display characters. Remove all characters.

LD 58 – Change the Translation Table Information.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	TBL	Translation Table access.
SNUM	0-15	System Number.
DNPS	xxxx yyyy	The DN to be translated and the number of the paging equipment to which the DN is assigned.
TABT	aaa	Table Type (Prompted when REQ = PRT)
RANG	xxxx...xxxx	Print DN Range from the first DN to the second DN (Prompted when REQ = PRT).

LD 18 – Define the ABCD table.

Prompt	Response	Description
REQ	NEW CHG	Add, or change 16 Button Data Block.
TYPE	ABCD	16 Button Data Block.

TBNO	1-254	Table Number.
DFLT	1-254	Default function table number.
PRED	(NO) YES	Pre-dial.
POST	(NO) YES	Post-dial.
CONT	(NO) YES	Control.

LD 18 – Define the pre-translation and post-translation list numbers.

Prompt	Response	Description
REQ	NEW CHG	Add, or change Pretranslation table assignment.
TYPE	PRE	Pretranslation calling group assignment.
CUST	xx	Customer number, as defined in LD 15
XLAT	0-254 0-8191	Pretranslation list (Calling group to Speed Call list correlation.)
	0-254 8191	If list number 8191 is assigned to a group, pretranslation is removed for that group.
-PRE	0-8190 X	Pre-translation Speed Call List number. Remove list.
-PST	0-8190 X	Post-translation Speed Call List number. Remove list.
-SDA	0-8190 X	Single-digit Access Speed Call List Number. Remove list.

Adding a Remote Radio Paging Flexible Feature Code

Task summary list

The following is a summary of the tasks in this section:

1 LD 87 – Define a Remote radio Paging (RRPA) FFC.

2 LD 11 – Configure the RPAG key for SL-1 sets.

3 LD 12 – Configure the RPAG key for Attendant Consoles.

LD 87 – Define a Remote radio Paging (RRPA) FFC.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
CUST	xx	Customer number, as defined in LD 15
FEAT	CDP	Coordinated Dialing Plan Feature.
TYPE	DSC TSC	Distant Steering Code. Trunk Steering Code.
DSC	xxxx	Distant Steering Code.
-FLEN	(0)-10	Flexible Length number of digits.
-DSP	LSC LOC DN	Display.
-RRPA	(NO) YES	Remote Radio Paging Access.
-RLI	xxx	Route List to be accessed for distant steering code.
-CCBA	(NO) YES	Collect Call Blocking.
TSC	xxxx	Trunk Steering Code.
-FLEN	(0)-16	Flexible Length number of digits.
-ITOH	(NO) YES	Inhibit Time Out option.

-CCBA	(NO) YES	Collect Call Blocking.
-RLI	xxx	Route List to be accessed for trunk steering code.

LD 11 – Configure the RPAG key for SL-1 sets.

Prompt	Response	Description
REQ:	CHG	Change RPAG key assignment.
TYPE:	aaa	Type of Data Block.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx aaa yyyy	To define an RPAG key with the RRPA FFC.

LD 12 – Configure the RPAG key for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	xxxx	Type of Attendant Console. xxxx = 1250, 2250
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx RPAG yyyy	To define an RPAG key with the RRPA FFC.

Feature operation

The following occurs when more than one RPS is configured per customer:

- The system number is transparent to the caller;
- The DN-PSA code translation table decides which RPS to use; and
- The trunk search is done after the DN is entered.

When one RPS is configured per customer, the trunk search is made after the FFC is entered.

Different call progress tones are provided by the RPS depending on the mode digit and state of the paging call.

Automatic pre-selection

Meridian 1 proprietary or SL-1 telephone

The following are the operation steps:

1 Off-hook.

- **a.** Set receives dial tone.

2 Enter the RPAC (FFC) for initiating RPA.

- **a.** Set receives paging tone if FFC is valid.
- **b.** Set receives CTVN treatment if FFC is invalid.
- **c.** Set receives congestion tone (as configured) if no trunk is available in a single system.

3 Enter the DN of party to be paged.

- **a.** Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.
- **b.** Set receives no tone from the system if speechpath is provided.
- **c.** Set receives CTVN treatment if DN is invalid.
- **d.** Set receives congestion tone if no paging trunks are available.
- **e.** Set receives busy tone if absence signal is received.

Attendant Console

When paging from a PSTN set, the attendant can access the RPA feature using the above steps and then transfer the call (similar to transferring to a normal set).

Automatic post-selection

Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

Meridian 1 proprietary sets

The following are the operation steps:

1 Off-hook.

- **a.** Set receives dial tone.

2 Enter the DN of party desired to be reached.

- **a.** Set receives ringback or busy tone if DN is valid.
- **b.** Set receives CTVN treatment if DN is invalid.

3 Press Recall key.

- **a.** Set receives recall signal.

4 Press single digit 0 - 9 for speed call list.

5 Press single alphabetic A - D, where character is a RPAG key (for RPA) for 16-Button DTMF set.

6 Enter RPAC (FFC) for initiating RPA.

- **a.** Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.
- **b.** Set receives no tone from the system if speechpath is provided.
- **c.** Set receives CTVN treatment if FFC or DN is invalid.
- **d.** Set receives congestion tone if no paging trunks are available.
- **e.** Set receives busy tone if absence signal is received.

SL-1 set

The following are the operation steps:

1 Off-hook.

a. Set receives dial tone.

2 Enter the DN of party to be paged.

a. Set receives ringback or busy tone if DN is valid.

b. Set receives CTVN treatment if DN is invalid.

3 While in ringing or busy state, press RPAG key (for RPA).

4 Press single digit 0 - 9 for speed call list.

a. Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

b. Set receives no tone from the system if speechpath is provided.

c. Set receives CTVN treatment if FFC or DN is invalid.

d. Set receives congestion tone if no paging trunks are available.

e. Set receives busy tone if absence signal is received.

Attendant Console

The following are the operation steps:

1 Off-hook.

a. Set receives dial tone.

2 Enter the DN of party to be paged.

b. Set receives ringback or busy tone if DN is valid.

c. Set receives CTVN treatment if DN is invalid.

3 Press RPAG key (for RPA).

a. Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

b. If the paging call recalls, the attendant can re-extend the call.

c. Set receives CTVN treatment if FFC or DN is invalid.

d. Set receives congestion tone if no paging trunks are available.

e. Set receives busy tone if absence signal is received.

Manual pre-selection

Meridian 1 proprietary sets

The following are the operation steps:

1 Off-hook.

a. Set receives dial tone.

2 Enter the RPAC (FFC) for initiating RPA.

a. Set receives paging tone if FFC is valid.

b. Set receives CTVN treatment if FFC is invalid.

c. Set receives congestion tone (as configured) if no paging trunk is available.

3 Enter the DN of party desired to be reached.

a. Set receives ringback or busy tone if DN is valid.

b. Set receives CTVN treatment if DN is invalid.

4 Enter mode digit.

5 Enter information to be sent.

6 Enter **#** for end of information.

a. Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

b. Set receives busy tone if absence signal is received.

Attendant Console

When paging from a PSTN set, the attendant can access the RPA feature using the above steps and then transfer the call (similar to transferring to a normal set).

Manual post-selection

Single-digit post-selection access codes are not supported at Remote Radio Paging (RRPA) nodes.

Meridian 1 proprietary set

The following are the operation steps:

1 Off-hook.

- **a.** Set receives dial tone.

2 Enter the DN of party to be paged.

- **a.** Set receives ringback or busy tone if DN is valid.
- **b.** Set receives CTVN treatment if DN is invalid.

3 Press Recall key.

- **a.** Set receives recall signal.

4 Press single digit 0 - 9 for speed call list.

5 Press single alphabetic A - D, where character is a RPAG key (for RPA) for 16-Button DTMF set.

6 Enter RPAC (FFC) for initiating RPA.

- **a.** Set receives no tone from the system if speechpath is provided.
- **b.** Set receives CTVN treatment if FFC or DN is invalid.
- **c.** Set receives congestion tone if no paging trunks are available.

7 Enter mode digit.

8 Enter information to be sent.

9 Enter **#** for end of information.

a. Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

b. Set receives busy tone if absence signal is received.

Attendant Console

The following are the operation steps:

1 Off-hook.

a. Set receives dial tone.

2 Enter the DN of party to be paged.

a. Set receives ringback or busy tone if DN is valid.

b. Set receives CTVN treatment if DN is invalid.

3 Press RPAG key.

a. Set receives ringback tone, call progress tones or silence (as configured) if paging was successful.

b. If the paging call recalls, the attendant can re-extend the call.

c. Set receives CTVN treatment if FFC or DN is invalid.

d. Set receives congestion tone if no paging trunks are available.

Answering the paging call

Paged party

The paged party receives a paging indication followed by one of the following types of information:

- no information
- a short speech cut-through, or
- digits displayed on receiving device.

A paged party can respond after receiving the information, as in the following:

- When the information is the caller's DN, the paged party responds by initiating a normal station-to-station call.
- When the information is not telephone related, the receiving device might get a coded message to perform some action.

Pre-selection and post-selection

Meridian 1 proprietary and SL-1 sets

The following are the operation steps:

1 Off-hook from any set on the system.

a. Set receives dial tone.

2 Enter the FFC for answering paging calls.

a. Set receives paging tone if the FFC is valid.

b. Set receives CTVN treatment if FFC is invalid.

3 Enter DN of your set.

a. Set is connected to the caller if the DN is valid.

b. Set receives CTVN treatment if the DN is invalid or is not being paged.

Attendant Console

When answering a paging call from a PSTN set, the attendant is required to make the connection. The attendant dials using the above method (FFC and DN) as if the call is being extended to another set.

Radio Paging Product Improvements

Contents

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Feature description

A Radio Paging system is a communications tool used to contact mobile parties by means of radio signals. A caller can use their telephone set to page a mobile party who has a mobile portable receiving device.

This product improvement enables RPA to recall the attendant who originated the Radio Paging call only; the attendant may be located anywhere within an ISDN Meridian Customer Defined Network (MCDN) configured with Network Attendant Services (NAS).

The improvement also enables an attendant's display to display paged name, instead of answering name, on the paging party when answered, and to make network Radio Paging show the same display information as in the standalone operation. For more information about Radio Paging, please see the Radio Paging feature module in this guide.

Operating parameters

Since ISDN Basic Rate Interface (BRI) sets do not support Flexible Feature Codes (FFCs), they cannot be used to access or answer RPA calls if the BRI sets are local on the paging node. For network situations, BRI sets can access and answer remote RPA calls. This is possible because the Radio Paging Access Code (RPAX)/Radio Paging Answering Code (RPAN) FFCs are dialed as Distant Steering Codes (DSCs)/Trunk Steering Codes (TSCs).

For Pre-selection Paging, if the paged DN following the RPAX FFC is not local to the paging node, the Call Party Name Display (CPND) name for this DN cannot be obtained to be displayed on the calling party's terminal. If the paged DN is local on the paging node and has CPND defined, the CPND can be retrieved and sent to the calling party for display purposes. For Post-selection Paging, the CPND of the paged DN will be displayed even if the DN is not local to the paging node.

If a network call comes in to a set on the paging node and is redirected to paging by Call Forward No Answer (CFNA), the calling name cannot be retrieved and updated on the answering set when the paging call is answered. This happens only if the set on the paging node has CPND defined. If the set does not have CPND defined, the calling name can updated on the answering party's set.

The following hardware is required for Radio Paging operation: Radio Paging System equipment meeting European Selective Paging Manufacturers' Association (ESPA) requirements; trunk cards (QPC296/QPC287/QPC551/QPC71/QPC237/NTD9742A/NT5K19AA) or Extended Flexible E&M (XFEM) cards (NT5K83/NT5K72/NT5K50/NT5K19).

The following hardware is required for Large Systems: PRI – NT5D97; DDP2 – NT8D72; and DCH – QPC757, NT6D11, or NT6D70 (MSDL).

The following hardware is required for Small Systems and Succession 1000 systems: PRI – NTAK09 with NTAK93 data port; PRI2 – NTAK79, or NTDK50 with NTBK51 DCHI data port; ISL – NTAK02.

Feature interactions

Call Detail Recording Enhancement

When an attendant makes an outgoing call (established on the source side) and then extends the call to remote radio paging on another node by using a normal trunk (for example, Trunk X), an “S” record is printed when the attendant releases to extend the call to network RPA.

If the outgoing trunk call releases before the paged call is answered, the “E” record will show the normal trunk ID (Trunk X).

If the paged call has been answered when the outgoing trunk call releases, the “E” record will show the paged DN instead of Trunk X.

Display of Calling Party Denied

If this feature is enabled (packaged under the International Supplementary features package 131), additional Classes of Service can be assigned to sets to determine whether or not their DN and CPND information will be displayed on other sets. No CPND or DN information is displayed on sets involved in a network RPA call that have name display denied or digit display denied Class of Service.

Network Attendant Services

Network Attendant Services (NAS) configuration is a requirement for the Network Radio Paging (NRPA) Recall to Same Attendant (RTSA) feature. Without NAS, NRPA RTSA is not active, and existing operation will be followed.

With NAS configured, if an RPA recall to the attendant on the originating node is not allowed, the recall will be presented on the paging node. Existing operation prior to this development is performed. There is no new interaction introduced with NAS features.

Slow Answer Recall Modification

With the Slow Answer Recall Modification (SLAM) feature enabled, when the attendant answers a recall the destination party is disconnected. This also applies to Radio Paging.

When the attendant answers a paging recall, the call is removed from the meet-me queue and the recall cannot be answered by the paging party by using RPA Answer. The paging party is put on the source side of the attendant; there is nothing connected on the destination side. The attendant cannot extend the call to paging by pressing the Release key. Pressing the Release key will disconnect the paging party from the source side and the attendant will become idle.

The attendant can extend the call to Radio Paging again by either: dialing the RPAX FFC + the DN (preselection); or dialing the DN, and while the DN is ringing or busy pressing the RPAG key (post-selection).

Feature packaging

Radio Paging (RPA) package 187 must be provisioned to activate this feature.

To gain access to RPA, Flexible Feature Codes (FFC) package 139 must be provisioned.

For the Radio Paging network recall operation, Network Attendant Service (NAS) package 159 must be provisioned.

For Remote Radio Paging, Coordinated Dialing Plan (CDP) package 59 is required to define RPA FFCs as Distant Steering Codes (DSCs) or Trunk Steering Codes (TSCs).

To display characters instead of the Radio Paging Flexible Feature Code, Calling Party Name Display (CPND) package 95 is required.

Integrated Services Digital Network (ISDN) package 145, and its dependencies, are required for operation in an MCDN ISDN network.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 87 – Set up remote Radio Paging on originating node.
2. LD 15 – In order for the Recall to Same Attendant portion of this feature to operate network wide, the Recall to Same Attendant (RTSA) prompt has to be activated on the originating node as follows:

LD 87 – Set up remote Radio Paging on originating node.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	TSC DSC	Trunk/Distant Steering Code (enter RPAX/RPAN FFC defined on paging node).
TSC DSC	xxxx	Radio Paging FFC from paging node.
RRPA	(NO) YES	Remote Radio Paging option.
RLI		Route List Index of route list block used to route to paging node.

LD 15 – In order for the Recall to Same Attendant portion of this feature to operate network wide, the Recall to Same Attendant (RTSA) prompt has to be activated on the originating node as follows:

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	ATT	Attendant console options.

...		
- RTSA	(RSAD) RSAA RSAX	Recall to same attendant denied. Recall to same attendant allowed. Recall to same attendant with queuing on busy.

Feature operation

With ISDN NAS enabled, the RPA Recall will recall to the same attendant who originated the call. The attendant may be located anywhere in the ISDN NAS network.

When the originating attendant answers the RPA recall, the call can be extended again by simply pressing the Release key.

When the paged party answers, recall to the originating attendant will be cancelled if the attendant has not yet answered.

If the paged party answers while the paging call is recalled to the originating attendant (buzzing), the request to cancel the recall is sent from the paging node to the originating node. If the attendant answers the recall before receiving the cancel message, the attendant is connected to both the paging and answering parties.

If the RPA RTSA network wide feature is not allowed, the recall is presented on the paging node. Existing operation prior to this development is performed. The RPA RTSA network wide feature is not allowed when one of the following conditions occurs:

- The originating attendant is busy (active on a loop) and RTSA is not RSAX on the originating node.
- The originating attendant is disabled or in maintenance mode.
- The originating attendant is in Night Service.
- The originating attendant is in Position Busy mode.
- The paging call was not handled by an attendant on the originating node. This includes:
  - — A set directly dials access to remote paging.
  - — The call is transferred by a set to remote paging.

— An attendant dials access to remote paging on the source side, with no other parties involved.

- The originating attendant never released to extend the paging call to the calling party (that is, the attendant has the calling set on the source side and the paging call on the destination side at recall time).

The recall time out for an RPA call is defined on the node that is directly connected to the RPA system, not the originating node from where the attendant extended the call. This is because the RPA timer is usually longer than the normal recall time out so that the paged party will have enough time to answer the call.

Radio Paging Product Improvement Continuation

Contents

This section contains information on the following topics:

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Feature description

A Radio Paging System (RPS) is a communications tool used to contact mobile parties by means of radio signals. With this system, a set can page a mobile party that is equipped with a radio paging device. The Radio Paging Product Improvement Continuation enhances the performance of the Radio Paging feature by providing the following:

- an increase in the number of digits sent to and displayed on a Radio Paging device
- the ability to activate/deactivate Pretranslation for Radio Paging calls
- five internal/external call treatments to a pager installed in the paging rack

Pager Display

With the existing Radio Paging functionality, when Calling Line Identification (CLID) information is sent to a paging device, a maximum of seven digits are displayed on the pager.

With the Radio Paging Product Improvement Continuation, however, up to 16 digits can be displayed on a pager. Therefore, it is possible for the entire CLID information to be displayed. In order to specify the number of digits (0-16) to be sent to the Radio Paging System, the Transmit Caller's DN (TRDN) prompt must be defined in LD 58.

Pretranslation

Pretranslation allows the creation of a flexible dialing plan by using Speed Call lists as Pretranslation Tables. With the Radio Paging Product Improvement Continuation, Pretranslation is activated/deactivated for Radio Paging calls by defining the Pretranslation (PRET) prompt in LD 58. This activation/deactivation takes place regardless of whether or not Pretranslation is allowed at a customer level.

Pagers installed in the paging rack

With existing Radio Paging functionality, the treatment of external calls forwarded to pagers in the paging rack is defined by the Recall if busy from Radio Paging (RCAL) prompt in LD 58. If RCAL is set to NO, the caller receives a busy tone. If RCAL is set to YES, the call is routed to the attendant. When an internal call is forwarded to a pager in the paging rack, the caller receives a busy tone.

With this Product Improvement Continuation, the user chooses what happens to internal/external calls forwarded to a pager in the paging rack. The treatment of these calls is defined by the Treatment for Internal Calls (INTR) and Treatment for External Calls (EXTR) prompts in LD 58.The INTR and EXTR prompts replace the RCAL prompt.

CAUTION

The treatment for external calls to a pager in the paging rack is **not** converted automatically. Therefore, the EXTR prompt must be defined. If EXTR is not defined, when an external call is forwarded to a pager in the paging rack, the call receives the default treatment for external calls (busy tone).

The Radio Paging Product Improvement Continuation offers the following five possibilities for the treatment of calls to pagers in the paging rack:

- The caller receives a busy tone.
- The call is routed to an attendant.
- The caller receives a special tone (SRC1-SRC8) or an announcement (with RAN equipment) delivered from the Tone and Digit Switch (TDS) card.
- The caller receives an announcement from a RAN machine.
- The call is routed to Meridian Mail.

Busy Tone

When INTR or EXTR is set to BUSY, the caller receives a busy tone.

Routed to an Attendant

When INTR or EXTR is set to ATT, the call is routed to an attendant.

Special Tone or Announcement

When INTR or EXTR is set to SRC1-SRC8, the caller receives a special tone, programmed in LD 56, or an announcement. After an announcement is provided to the caller, the call is disconnected. Recorded Announcement (RAN) equipment is required to provide this announcement.

Announcement from RAN

When INTR or EXTR is set to RAN, the caller receives an announcement from a RAN machine and is then disconnected or routed to an attendant after the message is heard. Post RAN treatment is defined by the RAN post announcement treatment (POST) prompt in LD 16.

For this enhancement to function, a RAN route must be specified by defining the Route number that provides the Recorded Announcement (RANR) prompt in LD 58. The RAN route must be specified prior to defining the RANR prompt.

Meridian Mail

When INTR or EXTR is set to MAIL, the call is routed to Meridian Mail. In this case, the caller receives an announcement stating that the call is being rerouted to Meridian Mail. With this enhancement, all Meridian Mail functions are available.

For this enhancement to function, the Meridian Mail Directory Number (MMDN) prompt must be defined in LD 58. Prior to defining the MMDN prompt, however, the Voice Automatic Call Distribution (ACD) messages queue must be defined in LD 23. The maximum input for Voice ACD is four digits or seven digits if the Directory Number Expansion (Seven Digit) (DNXP) package 150 is equipped.

Operating parameters

The Radio Paging Product Improvement Continuation is applicable on a stand-alone system with a Radio Paging system or in an Integrated Services Digital Network (ISDN) Meridian Customer Defined Network (MCDN) with a centralized Radio Paging System.

A maximum of 16 digits can be sent to Radio Paging equipment, as only 16 digits can be stored in the Calling Line Identification (CLID) field.

As per existing Radio Paging functionality, if the calling number is not available, the Route Access Code of the incoming trunk is displayed on the Radio Paging device.

If the calling number is shorter than the specified value defined at the TRDN prompt, the missing digits are replaced by zeros on the pager's display. With the existing functionality, a shorter calling number is also displayed on a pager in this manner.

If the calling number is greater than the specified value defined at the TRDN prompt, the most significant digits are displayed. The unnecessary digits are deleted.

The treatment of calls to a pager in the paging rack is only applicable if the Radio Paging device conforms to the standards of the European Selective Paging Manufacturer's Association (ESPA).

When the Recorded Paging Announcement (PANN) prompt is set to YES in LD 58, each redirected call to the paging equipment receives a recorded announcement stating that the called party is being paged. This announcement is provided even if the pager is in the paging rack.

When a pager is in the paging rack and PANN is set to YES, the caller receives an announcement stating that the pager is in the paging rack. After this announcement, the treatment, as a result of the INTR and EXTR prompts, is performed.

When the INTR or EXTR prompts are set to RAN and all Recorded Announcement (RAN) trunks are busy, the caller receives normal ringback tone. As soon as a RAN trunk becomes available, the caller hears a recorded announcement. This is as per existing RAN functionality.

Meridian Mail must be located on the same node as the paging device, in order for calls to a pager in the paging rack to be re-routed to Meridian Mail. If Meridian Mail and the paging device are not located on the same node, an error message appears at the overlay level.

When INTR or EXTR is set to Mail and the maximum number of calls to the Meridian Mail DN exceeds the limit that was set at the MAXP prompt in LD 23, the caller receives normal ringback tone. As soon as the number of calls is less than or equal to the MAXP value, the caller receives the recorded announcement or the defined Meridian Mail function. This is as per existing Meridian Mail functionality.

Feature interactions

Radio Paging Product Improvement Continuation has no specific interactions with existing features.

Feature packaging

The Radio Paging Product Improvement Continuation requires the following packages:

- Radio Paging (RPA) package 187, which requires the following package to access Radio Paging:
  - — Flexible Feature Codes (FFC) package 139
- Pretranslation (PXLT) package 92

The following packages are required for Meridian Mail:

- Make Set Busy (MSB) package 17
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package A (ACDA) package 45
- Command Status Link (CSL) package 77
- Command and Status Link with Alpha Signaling (CSLA) package 85
- Integrated Message System (IMS) package 35
- Message Waiting Center (MWC) package 46
- End-to-End Signaling (EES) package 10
- Directory Number Expansion (Seven Digit) (DNXP) package 150 for the Meridian Mail DN (MMDN) to contain a maximum of seven digits

The following package is required for Recorded Announcement (RAN):

- Recorded Announcement (RAN) package 7

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 58 – Allow or deny Pretranslation.

2 LD 58 – Set the internal and external treatment for calls to a pager in the paging rack, and set the number of digits of the caller's set transmitted to the paging equipment.

Note: The Radio Paging feature must be configured prior to implementing Radio Paging Product Improvement Continuation. If Pretranslation is to be allowed, the Pretranslation feature must also be configured. Depending upon how the INTR and EXTR prompts are defined, Mail and Recorded Announcement (RAN) must be implemented.

LD 58 – Allow or deny Pretranslation.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RPCD	Radio Paging Customer Data Block.
CUST	xx	Customer number, as defined in LD 15
...		
TRAN	 (TAB) TWO THR FOR NO	Translation type. Translation lookup table (default) Last two digits of DN Last three digits of DN Last four digits of DN No translation (DN sent as PSA code) The TRAN prompt is not given if MRPS = YES. TRAN is then forced to TAB.
- DNLN	0-(4)-16	DN length.
...		

RCTI	0-(30)-120	Time to wait for a “BUSY” transferring set to become idle. After this time, the call is routed to the attendant.
PRET	(YES) NO	Pretranslation for RPA calls (allowed) or denied.

LD 58 – Set the internal and external treatment for calls to a pager in the paging rack, and set the number of digits of the caller’s set transmitted to the paging equipment.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RPAX	Radio Paging Access Code Data Block.
CUST	xx	Customer number, as defined in LD 15
RPAX	nnnn	Radio Paging Access Code. This prompt is repeated to allow multiple entries. Access Codes must be previously defined in LD 57.
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
PANN	(NO) YES	Recorded Paging Announcement (denied) or allowed) for this route.
- RPAR	 0-511 0-127	Route number that provides the Recorded Announcement. For Large Systems For Small Systems and Succession 1000 systems
INTR	xxxx (BUSY) ATT SRC1-SRC8 RAN MAIL	Treatment for internal calls to a pager that is in the paging rack. Caller receives a busy tone (default). Call is routed to the attendant. Tones or announcement delivered from the TDS card which is programmed in LD 56. Call is routed to the RAN machine. Call is routed to Meridian Mail.
- RANR	 0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems

- MMDN	xxxx	Meridian Mail DN which provides the recorded announcement or the defined function. MMDN is prompted if INTR = MAIL. The MMDN may be up to four digits. However, if Directory Number Expansion (DNXP) package 150 is equipped, seven digits are allowed.
EXTR	xxxx (BUSY) ATT SRC1-SRC8 RAN MAIL	Treatment for external calls to a pager that is in the paging rack. Caller receives a busy tone (default). Call is routed to the attendant. Tones or announcement delivered from the TDS card, programmed in LD 56. Call is routed to the RAN machine. Call is routed to Meridian Mail.
- RANR	 0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems
- MMDN	xxxx	Meridian Mail DN which provides the recorded announcement or the defined function. MMDN is prompted if EXTR = MAIL. The MMDN may be up to four digits. However, if Directory Number Expansion (DNXP) package 150 is equipped, seven digits are allowed.
...		
OPER	(AUTO) MANU	Automatic operation (default). Manual operation.
- EXTM	(0)-9	External mode digit for this RPAX. EXTM is prompted when OPER = AUTO.
- INTM	(0)-9	Internal mode digit for this RPAX. INTM is prompted when OPER = AUTO.
- TRDN	(0)-16	Transmit the last x digits of the caller' s DN to the paging equipment. TRDN is prompted if OPER = AUTO.
...		

Feature operation

No specific operating procedures are required to use this feature.

Recall after Parking

Contents

This section contains information on the following topics:

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Operating parameters 495
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Feature implementation 496
Feature operation 497

Feature description

This enhancement to the Call Park feature causes a parked call to be recalled to the attendant or night DN if the attendant is in Night Service, rather than to the parking set, if not answered within a customer-defined period of time (two-minute maximum). The call may be external or internal.

Operating parameters

This feature does not apply to calls parked by Automatic Call Distribution (ACD) agents.

This feature operates in a standalone, but not in a network environment.

Feature interactions

Call Park

If the attendant is in Night Service, and a parked call is not answered within a customer-defined period of time (two-minute maximum), then the Recall after Parking feature recalls a parked call to the attendant or night DN instead of the parking telephone. If the parked call is recalled to a multiple appearance night DN, only the first appearance of the night DN will ring. The call may be external or internal.

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the night DN. If the night DN is busy, the call is queued if it is an external call.

Feature packaging

The Recall After Parking feature is included in Call Park (CPRK) package 33.

Feature implementation

LD 50 – Configure Recall after Parking at the RECA prompt.

Prompt	Response	Description
...		
CPTM	30-(45)-240	Call Park Timer (in seconds). The amount of time a call is held in the parked state before recalling the parking set or the attendant.
RECA	(NO) YES	Recall Attendant. YES = unanswered parked calls recall the attendant. NO = unanswered park calls recall the parking set.

Feature operation

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the Night DN. If the Night DN is busy, the external calls are queued.

Recall to Same Attendant

Contents

This section contains information on the following topics:

Feature description 499
Operating parameters 500
Feature interactions 500
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Feature operation 504

Feature description

The Recall to Same Attendant (RTSA) feature allows a recall to return to the attendant which last extended the call. If that attendant is busy, the recall is routed to either the first available idle attendant (option RSAA), or queued to the requested attendant until the attendant becomes idle (option RSAX). A call queued to an attendant in this way takes precedence over all other calls. Queued recalls are presented in the order in which they were queued.

The types of calls and recalls which can be queued are as follows:

- inter-attendant calls
- meter recalls
- slow answer recalls
- park recalls

- Camp-on recalls, and
- Call Waiting recalls.

Operating parameters

Attendant recalls brought about by switchhook flash, dial 0, call transfer, conference or the use of a recall key on a Meridian 1 proprietary telephone will not be affected by the RTSA feature.

RTSA will not apply to calls extended by Automatic Call Distribution (ACD) agents.

If an Attendant Console is maintenance or position busy, then recalls to it will be presented to the first idle Attendant Console, no matter which option has been specified.

If an attendant fails to answer a direct recall, that Attendant Console is forced into position busy, and the recall is presented to the first idle attendant.

RTSA is not supported by Centralized Attendant Service (CAS).

If the customer enters Night Service while recalls are timing for RTSA, these recalls will not be directed to the night station.

Feature interactions

AC15 Recall: Timed Reminder Recall

With the AC15 Timed Reminder Recall feature, if RTSA = RSAA the call is presented to the attendant who last extended the call, if RTSA = RSAX the call is presented to the attendant who last extended the call or put in the queue if this attendant is busy.

Attendant Forward No Answer

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call routed to the first available idle attendant.

Attendant Overflow Position

Recalls and inter-attendant calls are not routed to the Attendant Overflow Position.

Attendant Position Busy

If an Attendant Console is in maintenance or Position Busy when a Recall to Same Attendant call is recalled to it, the recall is presented to the first available idle attendant. If an attendant goes into Position Busy with a Return to Same Attendant call in Call Waiting, the waiting call is presented to the first available attendant.

Automatic Call Distribution

Recall to Same Attendant does not apply to calls extended by Automatic Call Distribution agents.

Call Forward No Answer

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call routed to the first available idle attendant.

Call Waiting Options

All options for call-waiting calls do not apply to calls queued to a specified attendant. The exception to this is the display call waiting key, which shows the number of calls in the overall attendant queue and the calls in the queue for a specified attendant.

Centralized Attendant Service

Centralized Attendant Service does not support the Recall to Same Attendant feature.

Flexible Attendant Call Waiting Thresholds

The Recall to Same Attendant (RTSA) feature has precedence over the Flexible Attendant Call Waiting Thresholds (FACWT) feature. If either RSAA or RSXA options are selected, RTSA has precedence over FACWT in determining the Call Waiting Lamp state. If one or more RTSA calls are waiting in the attendant queue, RTSA will set the Call Waiting Lamp state to wink (30 impulses per minute).

RTSA calls are not included when the FACWT feature determines the number of calls waiting.

Group Hunt

Calls redirected from a group hunt list via the listed DN or flexible attendant DN, and transferred back to the Pilot DN, are recalled if the Slow Answer Recall Timer expires. However, in practical configurations, the hunt terminates on the entry with the listed DN or attendant DN before the Slow Answer Recall Timer expires; consequently, the call is not redirected to that DN and presented on the applicable ICI key on the console. Therefore, the call is never presented as a recall, so that Recall to the Same Attendant does not apply.

Idle Extension Notification

An Idle Extension Notification recall will always recall to the same attendant, regardless of the configuration of the Recall to Same Attendant (RTSA) feature.

Multi-Party Operations

Users of analog (500/2500 type) telephones can perform an attendant recall during a two-party connection by performing a switchhook flash and then dialing the attendant DN.

Multi-Tenant Service

If a specified attendant is in maintenance or Position Busy, the recall first tries to terminate at another attendant within the same console group, and then to the night DN.

Network Attendant Service

This feature operates on a network-wide basis for the following call types:

- Slow Answer Recall
- Camp-on Recall, and
- Call Waiting Recall.

The operation of this feature is affected by the programming for the option in the Customer Data Block of the system where the attendant answering the call resides.

Periodic Pulse Metering

Meter recalls are returned to the same attendant whether Recall to Same Attendant is allowed or not. If Return to Same Attendant with Queuing on Busy (RSAQ) is selected as an option, the recalls are queued to a specified attendant.

Ring Again on No Answer

A telephone that is recalling the attendant cannot apply Ring Again on No Answer.

Tenant Service

Recall to Same Attendant applies to Tenant Service. If a specified attendant is in maintenance or Position Busy, the recall first tries to terminate at another attendant within the same console group, and then to the night DN.

Voice Messaging

Recall to Same Attendant does not apply to recalls from the Voice Messaging System.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Modify data for each customer member to be configured.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT	Attendant console options
...		
RTSA	(RSAD) RSAA	Recall to same attendant (denied) allowed.
	RSAX	Recall to same attendant allowed, with queuing on busy attendant.

Feature operation

If the requested attendant is idle, a recall to it will be presented on the loop key, and on the corresponding MTR, IAT, or RLL Incoming Call Indicator (ICI) key.

When a recall is queued specifically for an attendant, this will be indicated on the Attendant Console by a wink lamp state for the Call Waiting lamp.

Recall with Priority during Night Service

Contents

This section contains information on the following topics:

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Feature operation 506

Feature description

This feature (RPNS) places a priority level on the order in which calls queued to a Night DN are processed as follows:

- recall of an external call
- a new external call, and
- other calls.

This is the normal order during day processing.

Operating parameters

Due to the prioritizing of call processing, low priority calls may remain queued for a long time before being processed.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Configure Recall with Priority during Night Service.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	NIT	Night Service options
...		
- RPNS	(NO) YES	(Deny) allow Recall with Priority during Night Service.

Feature operation

The recall to the attendant appears on the Recall ICI key. If the attendant is in Night Service, the recall occurs to the Night DN. If the Night DN is busy, the external calls are queued.

If there is an occurrence of several calls of the same type to a station, the calls are presented to the station in their chronological order of arrival.

Recorded Announcement

Contents

This section contains information on the following topics:

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Feature packaging 509
Feature implementation 510
Feature operation 512

Feature description

The Recorded Announcement (RAN) feature allows the system to connect calls automatically to a customer-provided Recorded Announcement machine. Recorded Announcements can be used for:

- Automatic Call Distribution (ACD)
- Automatic Wake Up
- Intercept Treatment (INTR)
- Recorded Overflow Announcements (ROAs), and
- Network Queuing feature, which has Call Back Queuing (CBQ), Coordinated Call Back Queuing (CCBQ), Call Back Queuing to Conventional Main (CBQCM), and Off-Hook Queuing (OHQ).

The system software detects calls to connect to the Recorded Announcement (RAN) machine, determines the Intercept Treatment required, and connects the call to the proper Recorded Announcement. The system then monitors the RAN machine.

The system provides the software programs to control the announcement recorder and the circuit packs. Two types of circuit packs can be used:

- Recorded Announcement (RAN) Trunk Cards (QPC74) contain four identical trunk circuits for the interface between the system and the announcement machine. See *Circuit Card: Description and Installation* (553-3001-211) for engineering information. When the QPC74 is used, all ports on the card must be dedicated as TYPE RAN or TYPE MUS.
- Universal Trunk Cards (NT8D14AA) contain eight identical trunk circuits that can be configured independently in the system software. See *Circuit Card: Description and Installation* (553-3001-211) for a description.

Operating parameters

Dial access to RAN trunk groups is allowed and is limited only by Trunk Group Access Restrictions (TGARs).

When the QPC74 is used, all ports on the card must be dedicated as TYPE RAN or TYPE MUS.

Feature interactions

Conference
No Hold Conference

A RAN trunk cannot be Conferenced or No Hold Conferenced.

Collect Call Blocking

A RAN route is defined as having CCBA YES or NO, which is used if Coordinated Dialing Plan (CDP) or ACD queues were not used to get to the RAN route. If the call is routed through ACD/CDP to terminate on RAN, the Collect Call Blocking (CCB) treatment will depend upon the CCB data of the ACD/CDP, and not of the RAN route.

FCC Compliance for DID Answer Supervision

With FCC Compliance for DID Answer Supervision, incoming DID calls that are intercepted to a Recorded Announcement (RAN) are provided with answer supervision.

Group Hunt

Calls which are queued against the Group Hunt Pilot DN cannot receive Recorded Announcement.

Recovery on Misoperation of Attendant Console

If a Recorded Announcement is given to the destination side that has been intercepted, the connection to the destination side is considered as invalid. Therefore, if the attendant tries to extend the source to the destination using the RELEASE key or another LOOP key, the operation is ignored. The attendant must first press the RELEASE DESTINATION key to release the destination, and then extend the call to the source. If the HOLD key is pressed, the source party is put on hold and the Recorded Announcement is disconnected on the destination side.

Source Included when Attendant Dials

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Trunk Traffic Reporting Enhancement

The Trunk Seizure Option is not supported on RAN trunks.

Feature packaging

Recorded Announcement (RAN) package 7, which requires Intercept Treatment (INTR) package 11.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Enable Recorded Announcement (RAN) trunk route.

2 LD 14 – Enable Recorded Announcement (RAN) trunk.

LD 16 – Enable Recorded Announcement (RAN) trunk route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RAN	RAN trunks.
RTYP	CAP	Code-a-Phone recording device. Software allows announcements of up to 608 seconds.
	AUD	Audichron recording device (required when connecting to a Universal Trunk Card). Software allows announcements of up to 64 seconds.
	CK2	Cook Electric recording device. Software allows announcements of up to 64 seconds.
	DGT	Digital Recorders 213300 & 213400. Software allows announcements of up to 256 seconds.
	CON	NT7M series digital recorders. Software allows announcements of up to 608 seconds.
REP	1-15	Number of times the announcement repeats during each connection.

POST	ATT	Call is routed to attendant after specified number of repetitions (applies to Direct Inward Dial [DID] calls on Intercept).
	DIS	RAN is removed after a specified number of repetitions.
STRT	IMM	Call connects immediately to announcement.
	DDL	Call connects to announcement at the start of announcement.
ASUP	(NO) YES	Supervision (is not) or is required to inform the Central Office (CO) when the call is answered.
ACOD	xxx...x	Trunk route access code.
Note: All RAN route members must be removed before the route can be removed.		

LD 14 – Enable Recorded Announcement (RAN) trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RAN	RAN trunk data block.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems

Note: If a night table is used with Network Automatic Call Distribution (NACD), the FROA and FRT values in LD 23 need to be set for the Recorded Announcement feature. FROA should be "NO" and FRT should be four seconds greater than the last entry time of the night table.

Feature operation

No specific operating procedures are required to use this feature.

Recorded Announcement Broadcast

Contents

This section contains information on the following topics:

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Feature description

The Recorded Announcement Broadcast (RANBRD) feature expands the existing functionality of the Recorded Announcement (RAN) feature. Previously, the Recorded Announcement (RAN) feature used one-to-one connection between a calling party and a designated RAN trunk connected to a physical Recorded Announcement machine. Therefore, if four calling parties were receiving RAN treatment then four RAN trunks were occupied to provide this functionality.

The Recorded Announcement Broadcast feature eliminates the need for multiple cross-connections to provide recorded announcement. With this feature, multiple calling parties receive RAN treatment from one RAN trunk. Thus allowing a RAN trunk to simultaneously broadcast announcements to a maximum of 48 calling parties per RAN trunk. This expansion maximizes the usage of available RAN trunks.

This feature also introduces the following enhancements:

- Incremental Software Management limits
- RAN signalling capabilities
- Multi-Channel RAN Machine Types and Modes
- Message Staging Through Queuing Thresholds for Delay Dial Start/Stop RAN machines
- Music on Waiting
- Traffic Study Option

Each of the above enhancements are discussed in the sections that follow.

Incremental Software Management Limits

Two new Incremental Software Management (ISM) limits on Broadcast Routes and Broadcast Connections are introduced with this feature.

LD 22 is modified to print the new ISM information on RAN Broadcast connections that is introduced for the RAN Broadcast feature. The existing SLT command prints the ISM information for the system.

Customers can modify ISM parameters via keycode. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects ISM parameters.

To expand ISM limits, customers must order and install a new keycode. This installation is performed using the Keycode Management feature. All Keycode Management commands are executed in LD 143. For further information on keycode installation, please refer to *Large System: Upgrade Procedures* (553-3021-258).

For further information on ISM, refer to the “Incremental Software Management” in *Features and Services* (553-3001-306), Book 2 of 3.

Broadcast Routes

The ISM limit on broadcast routes is based on the number of broadcasting RAN routes available on a system. A new ISM header in LD 16 indicates ISM broadcasting RAN information for the system. This information is updated as each new RAN broadcasting route is configured by the customer. The upper ISM limit for broadcast routes is 511 for Large Systems and 127 for Small Systems. Table 38 shows the Broadcast RAN Route ISM information that is added to the header in LD 16.

Table 38
New Broadcast RAN Routes ISM Information in LD 16

RAN RTE	AVAIL: xx	USED: xx	TOT: xx

Broadcast Connections

The ISM limit on broadcast connections is based on the number of broadcast RAN connections available on the system. Additional broadcast RAN connections can be purchased incrementally. A new ISM header in LD 14 indicates ISM broadcasting RAN connections ISM information for the system. Table 39 shows the Broadcast RAN Connections ISM information that is added to the header in LD 14.

Table 39
New Broadcast RAN Connections ISM information in LD 14

TNS	AVAIL: xxxxx	USED: xxx	TOT: xxxxx
RAN CON	AVAIL: xxxx	USED: xxx	TOT: xxxx

As each new broadcasting RAN trunk is configured, the number of available broadcast connections is subtracted from the maximum number of broadcast connections to the RAN trunk. Any calling party that is listening to a recorded announcement through a broadcasting RAN trunk represents a broadcast connection.

The following scenario provides a detailed example of the new ISM limits that are applicable to this feature. Assume that a customer has an upper ISM limit of 5 broadcast RAN routes and an upper ISM limit of 240 broadcast connections. When the customer defines a new broadcast RAN route, the new number of available broadcast RAN is equal to the upper limit less 1, in this case that would be 4 broadcast RAN routes. When the customer configures 2 RAN trunks for the RAN route in LDs 14 and 16 broadcast connections to each trunk. The number of available broadcast connections is now equal to the upper limit less the number of configured broadcast RAN connections. So, in this scenario the customer has a total of 208 (240 -16-16=208) broadcast connections and a total of 4 broadcast RAN routes.

RAN Signaling

Immediate Start

With immediate start RAN signaling, the calling party is connected to the recorded announcement immediately. With this signaling, calling parties barge-in on the announcement. Therefore, the calling party can be connected to the announcement such as the beginning, middle or end.

The RAN Broadcast feature allows immediate start configuration the option of receiving Music On Hold to calling parties waiting for RAN treatment.

Delay Dial

With delay dial RAN signaling, the calling party is only connected at the start of a recorded announcement. With RAN Broadcast, calling parties can have the option of Music On Hold while waiting for the start of the announcement.

Multi-Channel RAN Machine Types and Modes

Multi-Channel corresponds to multiple RAN channels that can be configured within one RAN trunk route. In a Multi-Channel RAN route, each trunk has its own dedicated RAN channel on a physical RAN machine. Multi-Channel RAN routes do not support the cross connecting (daisy chains) of multiple trunk ports together so that several callers hear the same RAN message.

As an example in Multi-Channel RAN configuration, a Level Start/Stop Multi-Channel (MLVL) route could have trunk ports each configured with its own RAN channel. Each trunk could be assigned several RAN Broadcast connections. If the message is 15 seconds long, then queuing could be configured to start playing a message every 3 seconds.

The new multi-channel machine types - Continuous Mode Multi-Channel (MCON), Pulse Start/Stop Multi-Channel (MPUL) and Level Start/Stop Multi-Channel (MLVL) - are not linked to RAN machine or a given trunk. All trunks belonging to the RAN route are considered independent. RAN trunks and RAN machine channels are connected one to one. Accordingly, if one RAN trunk is detected as faulty then all other trunks are not impacted.

For these new RAN machine types, the maximum length of the recorded announcement is configured is two hours. The meaning of a ground signal received from the RAN machine (play or idle) is configured in LD 16. This prompt was previously only applicable to XFEM RAN trunks.

These new RAN machine types are applicable to broadcasting and non broadcasting RAN routes.

Recorded Announcement Broadcast supports two machine modes: Continuous and Start/Stop. Both modes support immediate start and delay dial configurations.

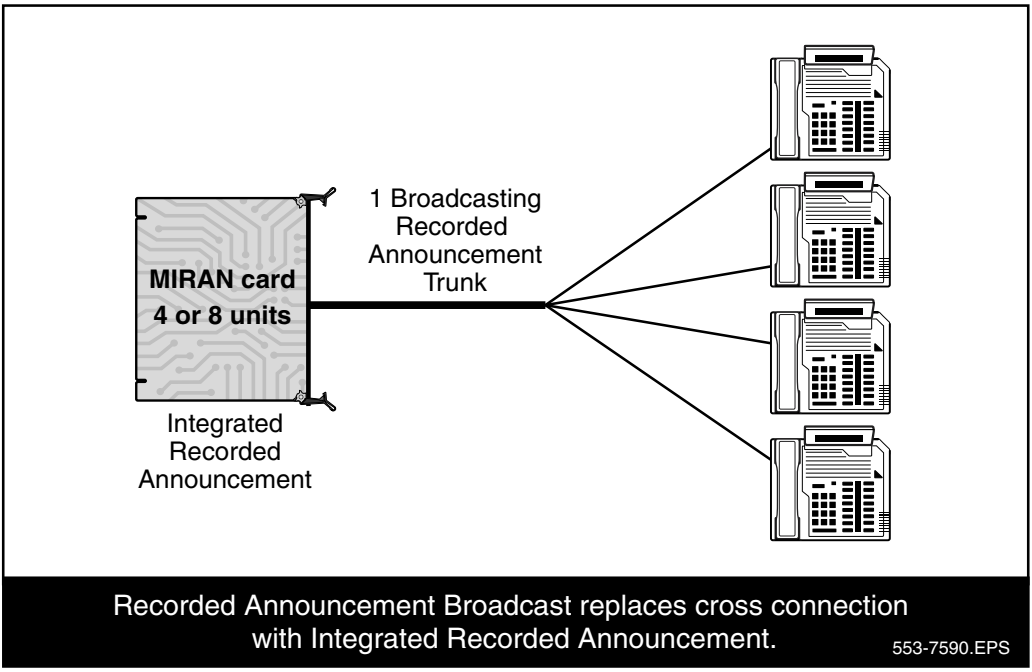
Table 40 outlines the hardware requirements and new RAN modes. RAN Broadcast requires an external RAN machine and a RAN trunk card.

Table 40
RAN modes and Hardware

Hardware	Types of RAN Modes		
	Continuous	Level Start/ Stop	Pulse Start/ Stop
QPC (X74)	X		X
XUT (NT8D14)	X		
EXUT (NT8D14)	X	X	X
XFEM (NT5K83)	X		X
MIRAN (NTAG36)	X	X	

As shown in Figure 8, the Meridian Integrated Record Announcement (MIRAN) cards eliminates the need for an external RAN machine. MIRAN emulates the Extended Universal Trunk (EXUT) card capabilities and provides built-in, physical RAN channels.

Figure 8
MIRAN Hardware



Continuous Mode

In Continuous mode, the recorded message is repeatedly played over and over. Calling parties requiring RAN treatment barge in on a playing message or receive ringback tone until the message starts over.

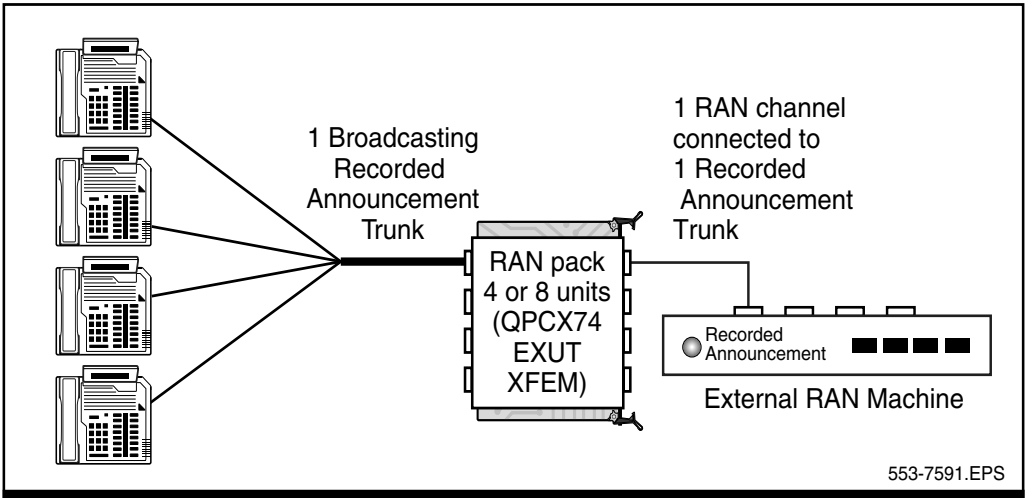
In Continuous mode, the maximum recommended amount of connections is between 10 and 16 connections per broadcasting RAN trunk. This amount depends on the following factors: CPU performance, answer supervision and delay between two announcements. This engineering requirement exists due to the fact that the system does not control the RAN channel. In Continuous mode, the message is continually running a recorded message with a short delay (usually less than 500 ms) between two announcements. If a RAN trunk is already broadcasting a recorded message to 12 calling parties and 12 calling parties require RAN treatment, then at the end of the message the system must disconnect these callers and connect the next calling parties before the message plays again.

Accordingly, the 12 connection limitation prevents the calling parties from hearing a RAN message that has already started playing. This value of 12 can be increased depending on the system specifications. As an example, a delay between two announcements that is greater than 500 ms. If answer supervision is not returned when the calling party connects to the recorded announcement, then up to 24 connections per Continuous mode RAN trunk are supported.

Recorded Announcement Broadcast introduces a new Continuous mode machine type called Continuous Mode Multi-Channel (MCON). Independent (asynchronous) RAN trunks can belong to a MCON RAN route which was not permitted with the existing Audichron/Cook 211 (AUD), NT7M Digital Recorders (CON) or 213300 and 213400 Digital Recorders (DGT) Continuous mode machine types.

Figure 9 illustrates RAN Broadcast using Start/Stop or Continuous Mode configuration.

Figure 9
RAN Broadcast using Start/Stop or Continuous Mode Configuration



With 1 Recorded Announcement Broadcast, 1 single Recorded Announcement Channel may broadcast up to 48 calling parties using one single RAN trunk.

Start/Stop Mode

In Start/Stop mode, the recorded message does not begin to play the recorded announcement until a start pulse signal is received from the RAN machine. There are two types of Start/Stop mode: Pulse Start/Stop and Level Start/Stop.

With Start/Stop configuration, the system controls when the RAN message starts and stops. Therefore, if 30 calling parties require RAN treatment, the system waits to start the recorded announcement until all 30 callers are ready to be connected. When the message is finished playing, the system disconnects all 30 callers and waits until the next 30 callers are queued before sending a message to the RAN machine to start playing the message. The recommended value for the maximum number of connections per broadcast start/stop trunk is 30. Again, this value can be increased depending on the system specifications. As an example, if the answer supervision signal is not returned when the calling party connects to the recorded announcement, then up to 48 connections per Start/Stop RAN trunk are supported.

With Pulse Start/Stop, the start signal is pulse. This pulse activates the playback of the recorded announcement. The announcement is played until completion. All other start pulses are ignored until the announcement has finished.

With Level Start/Stop, the start signal is a level. The leading edge of the start signal initiates the playback of the recorded announcement. This continues until either the trailing edge of the start signal occurs or the announcement has finished. When a trailing edge is detected, the recorded announcement is terminated and level start signal is sent to the RAN machine to immediately reset the recorded announcement.

Recorded Announcement Broadcast introduces two Start/Stop mode machines types called Pulse Start/Stop Multi-Channel (MPUL) and Level Start/Stop Multi-Channel (MLVL).

Message Staging

Recorded Announcement Broadcast allows the staging of recorded announcement for Delay Dial Start/Stop Machines. The staging of announcements is controlled by the queuing thresholds programmed in LD 16 for Delay Dial Start/Stop machines. With staging, if several copies of a recorded announcement are available on different RAN ports, then the start time of the recording can be staggered. For queued calling parties, this decreases the waiting time to hear the start of the announcement.

In Continuous modes, the staging of announcements is determined by the RAN machine.

Queuing Thresholds for Delay Dial Start/Stop Machines

The Recorded Announcement Broadcast feature introduces two new queuing thresholds for Start/Stop RAN machines configured with Delay Dial signaling (STRT=DDL in LD 16).

These new queuing thresholds allow customers to stagger recorded announcements using both time and number of calls as threshold triggers. Queuing thresholds optimizes a calling party's waiting time and the number of calls waiting to receive RAN treatment.

As an example, a customer has a recorded announcement that is 15 seconds in length. This announcement is used in a high volume Automatic Call Distribution (ACD) environment. In this scenario, a calling party requiring RAN treatment can range between 1 to 30 at any given time. With RAN Broadcast the 15 second message can be staggered. With this arrangement, 5 trunk ports could be configured in a RAN broadcast route with each trunk provisioned with 10 RAN broadcast connections. The message could then programmed to play every 3 seconds or when 10 caller are queued (TITH = 3 and NCTH = 10 in LD 16). In this configuration, each of the 5 trunks would be connected to individual RAN channels with each channel having the identical 15 second message. The calling party would only have to wait a maximum of 3 seconds before receiving a recorded announcement message.

With the new queuing thresholds, when the waiting or the number of calls threshold is met or exceeded the system searches for an available RAN Trunk and connects all queued callers waiting for a recorded announcement. If the system cannot locate an available trunk, then the waiting calls are requeued without a threshold so that waiting callers are connected to a RAN trunk as soon as it becomes available.

However, if RAN trunks are not available then callers are requeued without a threshold until the next RAN trunk is available. At this point, all threshold exceeded callers listen to the recorded announcement.

If no time or number threshold is configured, then all queued parties are connected to the first available RAN trunk. This includes callers that have just been queued by the system. Therefore, the system does not assign any priority to waiting callers when no thresholds have been configured.

Music on Waiting

Recorded Announcement Broadcast feature supports music on waiting for queued callers on both broadcasting and non-broadcasting RAN trunks. With this enhancement, music is provided when a calling party is queued to receive a recorded announcement. A selected music source is provided to waiting callers until the system locates an available RAN trunk. The music on waiting enhancement replaces ringback tone.

Traffic Study Option

The Traffic Period Option (TPO) allows a customer to enhance their TFC002 reports to accumulate trunk usage data after every traffic period instead of accumulating usage only after a call disconnects. With this option enabled in LD 17, the Common Channel Signaling (CCS) associated with lengthy calls is reported in each traffic report interval throughout the duration of the call.

Previously, this feature did not apply to RAN and Music trunks. However, with the introduction of the RAN Broadcast feature, changes are made to the Trunk Traffic Reporting Enhancement with the introduction of TFC111. The TFC111 report provides information on the usage of broadcasting routes.

For the TFC111 to be output, the customer report number 11 must be selected using the SOPC command in LD 2. For example, for Customer 0, SOPC 0 11 is entered. To print the TFC111 report, the TOPC command in LD 2 is used. For example, for Customer 0, TOPC 0 11 is entered. The TFC 111 report is also printed when automatic traffic reports are scheduled in LD 2.

A traffic message is output each time the number of active broadcasting connections is equal to the system's ISM limit.

The new TFC111 report provides the following information:

- the trunk type
- the number of successful broadcast connections of the trunk associated with route
- the average duration of broadcast connects for route
- the average waiting time for RAN requests
- the maximum waiting time for RAN requests
- the waiting time threshold peg count
- the number of waiting parties threshold peg count
- the broadcast connection peg count for three lowest usage trunks

Table 41 is an example of the customer report, TFC 111, for RAN Broadcast routes.

Table 41
New Customer Traffic Measurement Outputs

System ID 0200	TFC111	
Customer Number 000		
Route Number 031	Trunk Type RAN	
Successful broadcast connections peg count 000817	Average call duration 00006	Average waiting duration 00004
Maximum waiting time 00007	Waiting time threshold peg count 00000	Number of waiting parties threshold peg count 00000
Broadcast connections peg count for lowest trunk usage 00000	Broadcast connections peg count for next to lowest trunk usage 00000	Broadcast connection peg count for second lowest trunk usage 00002

Maximum number of connections per broadcasting RAN trunk

Table 42 shows the maximum number of connections per broadcasting RAN trunk that can be configured. These values depend on system configuration; therefore, some systems can allow greater values or request lower values.

When no answer supervision signal is to be returned at the time the caller receives the announcement, more connections are supported. This is the case with unsupervised trunks, internal calls, or when the answer signal has already been sent.

If answer supervision is returned, there is a high impact on real-time. Therefore, it is recommended that the maximum number of connections per RAN trunk be set to a lower value (See Table 42).

To achieve maximum efficiency, TFC111 and the TITH and NCTH thresholds can be used. For instance, the difference between the number of times TITH was met and NCTH was met provides an indication of how the system reacts to the incoming RAN request rate. In the case of a high rate, a greater number of NCTH was met than TITH. This indicates that the number of connections is insufficient.

Table 42
Recommended maximum number of connections per trunk

RAN mode	Is answer supervision returned when RAN is provided?	Recommended maximum number of connections per RAN trunk
Continuous mode with less than 500ms between two announcements	Yes	up to 12
Continuous mode with less than 500ms between two announcements	No	up to 24
Start/Stop mode	Yes	up to 30
Start/Stop mode	No	up to 48

Operating parameters

The Recorded Announcement Broadcast feature is applicable to RAN routes only.

The Meridian Integrated Recorded Announcement (MIRAN) card provides a multi-tasking environment for certain voice processing intensive applications, such as RAN and Music on Hold. This card stores recorded music and announcements in flash memory using two audio ports. The setup or modification of sound files is done using a set or a TTY. This card stores recorded music and announcements in flash memory or PCMCIA flash memory cards. Music can be played from an analog source, such as a Compact Disc (CD) player or a Muszac source, through the MIRAN card. It is not a requirement that Music be recorded within MIRAN. The card plays music from other sources.

When configuring this feature, the mode supported by the external RAN machine and system hardware must match. The EXUT card supports continuous, pulse start/stop and level start/stop. The XFEM card supports continuous and pulse start/stop modes. The MIRAN card supports continuous and level start/stop modes.

The MIRAN card is associated with a certain port on the EXUT card. Each recorded announcement can be associated with more than one port at one time.

Traditional Recorded Announcement and Recorded Announcement Broadcast can exist on the same system.

If using a Start/Stop RAN machine, it is recommended that both the Waiting Time Threshold (TITH prompt) and the Number of Calls Waiting Threshold (NCTH prompt) be configured.

The Waiting Time Threshold (TITH) and the Number of Calls Waiting Threshold (NCTH) prompts should be configured to minimize caller's waiting time. TITH should be set to the length of the RAN message divided by the number of RAN trunks. NCTH should be set to the maximum number of connections per trunk divided by the number of RAN repetitions. All RAN trunks should have the same number of allowed connections to trigger RAN starts.

The continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel all support independent RAN trunks.

A RAN route can be modified to disallow broadcasting, provided that all trunks do not have any active calls connected when changes are made. When modifying a RAN route to allow broadcasting, the number of available ISM RAN connections must be sufficient or the change is not permitted.

In customer situations with high RAN usage, continuous RAN is recommended. In situations with a fluctuating or low incoming rate, a start/stop RAN with thresholds configured at a low value is recommended.

Feature interactions

Answer Supervision

Answer Supervision is provided based on the configuration of the RAN route. When music is provided to queued callers waiting for an announcement, the answer supervision is returned as though the recorded announcement was given.

Automatic Call Distribution Recorded Overflow Announcement

Automatic Call Distribution (ACD) and Recorded Overflow Announcement (ROA) allows queued calls to an ACD agent or attendant to be routed to a recorded announcement informing the calling party of the delay. If music is selected between the first and second recorded announcement, queued calls can be routed to a second announcement if they are still waiting in the queue.

When Music on Waiting is configured for the second RAN route, the music source selected by the Automatic Call Distribution or Recorded Overflow Announcement feature, already provided to a queued call, is not replaced by the one selected by the second RAN route when this queued call is waiting to be connected to the second RAN.

Automatic Wake Up

Automatic Wake Up (AWU) broadcast capability is independent of the RAN broadcast capability. AWU broadcast is only applicable to AWU trunks.

Incremental Software Management

The Incremental Software Management (ISM) limits introduced by this feature impact the number of units available and used by the ISM feature. The ISM header at the start of LD 14 is updated to indicate the broadcast RAN connections ISM information on the system.

INIT ACD Queue Call Restore

ACD calls queued for receiving RAN are restored by the INIT ACD Queue Call Restore feature following system initialization. All other calls queued for RAN are dropped, and the callers hear silence.

If system initialization occurs when an Automatic Call Distribution (ACD) call is being greeted by ACD RAN, the RAN greeting is automatically disconnected. If the call is restored by the INIT ACD Queue Call Restore feature, the call is presented to the appropriate ACD Directory Number as a new call.

When system initialization occurs, Music on Waiting is stopped and the restored call is presented to the ACD DN as a new call.

Integrated Call Center Management

Integrated Call Center Management (ICCM) broadcast capability is independent of the RAN Broadcast capability. ICCM broadcast is only applicable to IVR voice ports.

The script command GIVE RAN<RAN route number> connects a call to the specified RAN route and the RAN broadcast feature will apply if applicable.

Music Broadcast

The Music Broadcast feature is applicable to Music only, and the RAN Broadcast feature is applicable to RAN only.

Feature packaging

The Recorded Announcement Broadcast (RANBRD) feature is package 327. The following packages are also required:

- Recorded Announcement (RAN) package 7
- Intercept Treatment (INTR) package 11

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Define Continuous RAN Route.

2 LD 16 – Define Immediate Start/Stop RAN Route.

3 LD 16 – Define Delay Dial Start/Stop RAN Route.

4 LD 14 – Define new RAN Trunk.

5 LD 16 – Define Continuous RAN route with Meridian Integrated RAN (MIRAN).

6 LD 16 – Define Immediate Start/Stop RAN route with Meridian Integrated RAN (MIRAN).

7 LD 16 – Define Delay Dial Start/Stop RAN route with Meridian Integrated RAN (MIRAN).

8 LD 14 – Define a RAN trunk.

The following scenario provides details on how to configure RAN Broadcasting and Non Broadcasting using different applications such as Automatic Call Distribution (ACD) queues and intercept treatments.

Assume the following scenario exists. You have a system configured with non-broadcasting RAN. Your system has 3 RAN routes. Route 1 has 1 trunk with low usage and handles RAN intercept treatments. Route 2 has 8 trunks with variable usage and handles Recorded Overflow Announcement (ROA). Route 3 has 16 trunks with high usage and handles all Automatic Call Distribution (ACD) greetings into your call centre.

Table 43 and Table 44 provide a non-broadcasting and a broadcasting scenario respectively.

Table 43
Non-Broadcasting Scenario

RAN Routes	Usage	RAN Mode	Number of Trunks	RAN Machine Type
1	Low	Start/Stop	1	Cook 201/ QAY1.
2	Varied	Continuous	8	Audichron/ Cook 211 (required for XUT trunks)
3	High	Continuous	16	Audichron/ Cook 211 (required for XUT trunks)

In the non-broadcasting scenario the following system requirements exist:

- a total of 25 (1 + 8 + 16) RAN trunks
- a total of 3 RAN channels

When using the RAN Broadcast feature in the same scenario, RAN trunks and RAN channels requirements are reduced. With this feature, each group of RAN trunks is replaced by one broadcast RAN trunk with maximum number of connections set to the number of cross connected trunks. RAN Broadcast allows a maximum of 48 connections per RAN trunk.

Table 44
Broadcasting Scenario

RAN Routes	Usage	RAN Mode	Number of Trunks	RAN Machine Type	Broadcast Connection/ Trunk
1	Low	Start/Stop	1	Cook 201/ QAY1	non broadcast
2	Varied	Continuous	1	Audichron/ Cook 211 (required for XUT trunks)	8 connections
3	High	Continuous	1	Audichron/ Cook 211 (required for XUT trunks)	16 connections

In the broadcasting scenario, the following system requirements exist:

- a total of 3 (1+1+1) RAN trunks
- a total of 3 RAN channels
- a total of 2 RAN Broadcast Route ISM limits
- a total of 24 (8 + 16) RAN Connections ISM limits

The broadcasting scenario can be further enhanced if RAN routes 2 and 3 used a Delay Dial Start/Stop RAN trunk with the Number of Calls Waiting Threshold and Waiting Time Threshold configured.

LD 16 – Define Continuous RAN Route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RAN	Recorded Announcement trunk type.
RTYP	 AUD CON DGT MCON	Recording devices for RAN trunks where: Audichron/Cook 211 (required for XUT trunks). NT7M Digital Recorders. 213300 and 213400 Digital Recorders. Continuous mode Multichannel.
REP	1-15	Number of repetitions of recorded announcements.
POST	 ATT DIS	RAN Post announcement treatment where: Route to attendant after maximum repetitions Disconnect after maximum repetitions.
STRT	 IMM DDL	Start arrangement where: Immediately connect call to recording. Delay call connection until start of recording.
WAIT	RGB	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	 0-511 0-127	Music route for RAN queuing. For Large Systems For Small Systems and Succession 1000 systems MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow RAN broadcast for this route. NO = Deny RAN broadcast for this route (default).

ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Immediate Start/Stop RAN Route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RAN	Recorded Announcement trunk type.
RTYP	 CAP CKM PUL LVL MPUL MLVL	Recording devices for RAN trunks where: Code-A Phone. Cook 201 multichannel. Pulse start/stop (Enhanced Universal Trunk cards). Level start/stop (Enhanced Universal Trunk cards). Pulse start/stop multichannel. Level start/stop multichannel.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	IMM	Immediately connect call to recording.

WAIT	RGB	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	 0-511 0-127	Music route for RAN queuing. For Large Systems For Small Systems and Succession 1000 systems MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. NO = Deny broadcast capability for this route (default).
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Delay Dial Start/Stop RAN Route.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RAN	Recorded Announcement trunk type.

RTYP		Recording devices for RAN trunks where:
	CAP	Code-A Phone.
	CK2	Cook 201/QAY1.
	CKM	Cook 201 Multichannel.
	PUL	Pulse start/stop (Enhanced Universal Trunk cards).
	LVL	Level start/stop (Enhanced Universal Trunk cards).
	MPUL	Pulse start/stop multichannel.
	MLVL	Level start/stop multichannel.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	DDL	Delay call connection until start of recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT		Music route for RAN queuing
	0-511	For Large Systems
	0-127	For Small Systems and Succession 1000 systems MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
- TITH	(0)-300	Waiting Threshold in seconds. Default value of zero means no threshold applies.
- NCTH	(0)-100	Number of Calls Waiting Threshold. Default value of zero means no threshold applies.
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 14 – Define new RAN Trunk.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RAN	Recorded Announcement trunk data block.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
XTRK	a...a	Extended Trunk. To specify hardware, according to the RAN mode defined in LD 16, refer to Table 40.
RTMB	0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
- CONN	(4)-48	Define the maximum number of broadcast connections allowed for this trunk. ***Note:*** *CONN is only prompted for associated RAN route with broadcasting allowed (BDCT=YES in LD 16).*

Note: The following feature implementation is applicable to customers using the Meridian Integrated Recorded Announcement (MIRAN) card.

LD 16 – Define Continuous RAN route with Meridian Integrated RAN (MIRAN).

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems

TKTP	RAN	Recorded Announcement trunk type.
RTYP	MCON	Continuous Multi-channel.
- LGTH	4-(60)-7200	Maximum message length in seconds. This is only prompted for the continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel.
- GRD	(IDLE)	Ground signal from RAN indicates that machine is idle (default). PLAY = Ground signal from RAN indicates that machine is playing.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	aaa	Start arrangement where: IMM = Immediately connect call to recording. DDL = Delay call connection until start of recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	0-511 0-127	Music route for RAN queuing For Large Systems For Small Systems and Succession 1000 systems MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Immediate Start/Stop RAN route with Meridian Integrated RAN (MIRAN).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RAN	Recorded Announcement trunk type.
RTYP	MLVL	Level start/stop multichannel recording devices for RAN trunks.
- LGTH	4-(60)-7200	Maximum message length in seconds. This is only prompted for the continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel.
- GRD	(IDLE)	Ground signal from RAN indicates that machine is idle (default). PLAY = Ground signal from RAN indicates that machine is playing.
REP	1-15	Repetitions of recorded announcements.
POST	aaa	Post RAN treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	IMM	Immediately connect call to recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	 0-511 0 -127	Music route for RAN queuing For Large Systems For Small Systems and Succession 1000 systems MRT is only prompted for RAN route with WAIT = MUS.

BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 16 – Define Delay Dial Start/Stop RAN route with Meridian Integrated RAN (MIRAN).

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	RAN	Recorded Announcement trunk type.
RTYP	MLVL	Level start/stop multichannel recording devices for RAN trunks.
- LGTH	4-(60)-7200	Maximum message length in seconds. This is only prompted for the continuous mode multichannel, the level start/stop multichannel and the pulse start/stop multichannel.
- GRD	(IDLE)	Ground signal from RAN indicates that machine is idle (default). PLAY = Ground signal from RAN indicates that machine is playing.
REP	1-15	Repetitions of recorded announcements.

POST	aaa	RAN Post announcement treatment where: ATT = Route to attendant after maximum repetitions DIS = Disconnect after maximum repetitions.
STRT	DDL	Delay call connection until start of recording.
WAIT	(RGB)	Provide ringback for call queuing for RAN trunk (default). MUS = Provide music for calls queuing for RAN trunk.
- MRT	 0-511 0-127	Music route for RAN queuing For Large Systems For Small Systems and Succession 1000 systems MRT is only prompted for RAN route with WAIT = MUS.
BDCT	YES	Allow broadcast capability for this route. (NO) = Deny broadcast capability for this route (default).
- TITH	(0)-300	Waiting Time Threshold in seconds. The default value of (0) means no threshold applies. TITH is only prompted when BDCT = YES and STRT = DDL.
- NCTH	(0)-100	Number of Calls Waiting Threshold. Default value of zero means no threshold applies. NCTH is only prompted when BDCT = YES and STRT = DDL.
ASUP	(NO)	Do not return answer supervision (default). YES = Return answer supervision. CO = Return answer supervision only if originator is a Central Office trunk.
ACOD	x...x	Access Code for the trunk route. The Access Code must not conflict with the numbering plan. ACOD can be four digits, or seven digits with DNXP package 150 equipped.

LD 14 – Define a RAN trunk.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.

TYPE	RAN	Recorded Announcement trunk data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
XTRK	EXUT	Enhanced Extended Universal Trunk card. To use the new MIRAN card, the XTRK prompt must be set to EXUT.
...		
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
- CONN	(4)-48	Maximum number of broadcast connections allowed for this trunk. ***Note:*** CONN is only prompted for associated RAN route with broadcasting allowed (BDCT=YES in LD 16). ***Note:*** The CONN prompt defines the maximum number of broadcast connections allowed for a RAN trunk at any given time. As an example, if sixteen is configured, then the physical broadcasting trunk may broadcast up to sixteen callers at one time.

Feature operation

No specific operating procedures are required to use this feature.

Recorded Overflow Announcement

Contents

This section contains information on the following topics:

Feature description 543
Operating parameters 544
Feature interactions 545
Feature packaging 545
Feature implementation 546
Feature operation 549

Feature description

Recorded Overflow Announcement (ROA) allows delayed calls to the attendant to be connected to a recorded announcement notifying the calling party of the delay. A second recorded message can also be provided to the calling party repeatedly until an attendant answers the call.

A call that is waiting in the queue receives the first recorded message after the expiration of a timer (T1). After the message is given, the call returns to the attendant queue. While the call is in the waiting state, it can be connected either to Music (MUS), Ringback tone (RGB), or Silence (SIL).

If a second recorded announcement is specified, the call receives the message upon expiration of a second timer (T2). After the second message is given, the call is placed in the attendant queue again. There is no limit to the number of times a call can be given the second recorded message.

Operating parameters

Recorded Overflow Announcement (ROA) treatment is provided to call types assigned to Incoming Call Indicator (ICI) keys on the Attendant Console.

A maximum of 20 ICI keys can be assigned to receive Recorded Overflow Announcement (ROA) treatment.

The delay time thresholds for the first and second recorded announcements (T1 and T2) are assigned in LD 15. The thresholds shown in Table 45 can be defined for these timers.

Table 45
Delay time thresholds

	Thresholds		
	Minimum	**Default**	**Maximum**
T1	0 seconds	20 seconds	2,044 seconds
T2	2 seconds	40 seconds	2,044 seconds

Loop start trunks do not provide disconnect supervision and are not recommended for use with the ROA feature. A call on a loop start trunk that is abandoned after the recorded message is given must be manually cleared by the attendant.

ROA is not provided on release link trunks from Centralized Attendant Service (CAS) remote locations.

When the CAS feature is activated at a remote circuit switched network, the ROA feature is inactive at the remote site.

If music is required, the Music (MUS) package 44 must be equipped. Music can be provided after the first and second Recorded Announcement (RAN). A customer provided music source is required, connected through a Music trunk. Music is provided to delayed calls through a conference circuit pack in a listen-only mode. The music source provided by the customer must be compatible with the RAN trunk card.

Private Lines are not eligible for ROA.

ROA is not provided for any type of transferred call. A recalled call from Meridian Mail, an analog (500/2500 type) telephone, or a proprietary telephone, will not be eligible for ROA treatment.

ROA is only provided for call types assigned to Incoming Call Indicator (ICI) keys. The following call types are eligible, if related ICI keys are assigned:

- Trunk routes
- LDN 0 through LDN 3
- Dial 0
- Dial 0 Fully Restricted
- Intercept Treatment (INTR)
- Call Forward Busy
- Call Forward No Answer
- Message Waiting (MW)
- Lockout, and
- Station Category Indication (SCI).

Feature interactions

Automatic Call Distribution (ACD)

The RAN route used for ROA can be the same route that is used for ACD and Intercept Treatment.

Call Transfer

ROA is not provided for any type of transferred call.

Night Service

The ROA feature is inactive when the system is in Night Service.

Feature packaging

Recorded Overflow Announcement (ROA) package 36 requires Recorded Announcement (RAN) package 7.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 16 – Enable Recorded Announcement (RAN) trunk route..

2 LD 14 – Enable Recorded Announcement (RAN) trunk.

3 LD 15 – Configure Recorded Announcement (RAN) in the customer data block..

LD 16 – Enable Recorded Announcement (RAN) trunk route.

Prompt	Response	Description
REQ	NEW, CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
....	....	
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
....	....	
TKTP	RAN	RAN trunks.

RTYP	CAP	Code-a-Phone recording device. Software allows announcements of up 608 seconds.
	AUD	Audichron recording device (required when connecting to a Universal Trunk Card). Software allows announcements of up to 64 seconds.
	CK2	Cook Electric recording device. Software allows announcements of up to 64 seconds.
	DGT	Digital Recorders 213300 & 213400. Software allows announcements of up to 256 seconds.
	CON	NT7M series digital recorders. Software allows announcements of up to 608 seconds.
....	....	
REP	1-15	Number of times the announcement repeats during each connection.
POST	ATT	Call is routed to attendant after specified number of repetitions (applies to Direct Inward Dial [DID] calls on Intercept).
	DIS	RAN is removed after a specified number of repetitions (call is kept in Automatic Call Distribution queue).
STRT	IMM	Call connects immediately to announcement.
	DDL	Call connects to announcement at the start of announcement.
....	....	
ASUP	YES	Return Answer Supervision by RAN to originator. ASUP=NO (Default) ***Note:*** ASUP must be set to YES to allow the following options in LD 15 (at the WAIT prompt): Caller hears Ringback (RGB), Music (MUS), or Silence (SIL) while waiting.
ACOD	xxx...x	Trunk route access code.
....	....	

Note: All RAN route members must be removed before the route can be removed.

LD 14 – Enable Recorded Announcement (RAN) trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RAN	RAN trunk data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems

LD 15 – Configure Recorded Announcement (RAN) in the customer data block.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ROA	Recorded Overflow Announcement options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
OPT	(ROX), ROI	Recorded Overflow (excluded) included.
- FRRT	xxx	Route number for the first recorded announcement.
- FRT	0-(20)-2044	Time in seconds before the first announcement plays.
- SRRT	xxx	Route number for the second recorded announcement.
- SRT	2-(40)-2044	Time in seconds before second announcement plays.
- WAIT	RGB, MUS, SIL	Caller hears Ringback (RGB), Music (MUS), or Silence (SIL) while waiting.

- - MURT	xxx	Route Number for Music route if WAIT = MUS.
- RICI	xx . .xx . .xx	Incoming Call Indicator (ICI) key numbers eligible for ROA.

Feature operation

No specific operating procedures are required to use this feature.

Recorded Telephone Dictation

Contents

This section contains information on the following topics:

Feature description 551
Operating parameters 551
Feature interactions 552
Feature packaging 552
Feature implementation 552
Feature operation 553

Feature description

This feature provides dial access to customer-supplied dictation equipment. Operation of the equipment can be either voice or dial controlled. The actual controls vary with the type of dictation equipment used.

To access the dictation equipment, the user dials the access code assigned to the dictation route. Access to the route is controlled by Trunk Group Access Restrictions (TGARs).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Multi-Party Operations

Users of analog (500/2500 type) telephones cannot make a consultation call while connected to a dictation trunk.

Conference

Recorded Telephone Dictation trunks cannot be used in a conference call.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 16 – Enable a trunk route for the Recorded Telephone Dictation feature.
2. LD 14 – Enable a trunk for the Recorded Telephone Dictation feature.

LD 16 – Enable a trunk route for the Recorded Telephone Dictation feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	DIC	Recorded Telephone Dictation trunk route.

ICOG	OGT	Outgoing trunk route.
ACOD	xxx...x	Directory Number (DN) to dial to access the dictation device.

LD 14 – Enable a trunk for the Recorded Telephone Dictation feature.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
SIGL	aaa	Trunk signaling.
STRO	aaa	Outgoing start arrangement.
SUPN	(NO) YES	Answer and disconnect supervision (not) required.

Feature operation

No specific operating procedures are required to use this feature.

Recovery on Misoperation of Attendant Console

Contents

This section contains information on the following topics:

Feature description 555
Operating parameters 555
Feature interactions 556
Feature packaging 557
Feature implementation 557
Feature operation 557

Feature description

The Recovery of Misoperation on the Attendant Console feature provides a safeguard in the system software that prevents calls from being inadvertently disconnected.

Operating parameters

For Centralized Attendant Service, misoperation of the Attendant Console at the main node cannot be prevented.

Feature interactions

Call Forward All Calls
Call Forward Busy
Call Forward by Call Type
Call Forward External Deny
Call Forward, Internal Calls
Call Forward No Answer
Call Forward No Answer, Second Level
Hunting

These features take precedence over the Recovery of Misoperation feature.

Electronic Switched Network

If the attendant dials an incomplete Electronic Switched Network (ESN) number as a destination, pressing the Release key or another loop key is ignored. The attendant can dial more digits as long as the interdigit timer has not timed out. To dial to another number, the attendant must first press the Release Destination key to release the destination.

Music on Hold

Music on Hold, if allowed, is applied to calls put on hold due to the Autohold on the loop key option.

Recorded Announcement

If a recorded announcement is given to the destination side that has been intercepted, the connection to the destination side is considered as invalid. Therefore, if the attendant tries to extend the source to the destination using the Release key or another loop key, the operation is ignored. The attendant must first press the Release Destination key to release the destination, and then extend the call to the source. If the Hold key is pressed, the source party is put on hold and the recorded announcement is disconnected on the destination side.

Through Dialing

If an attendant dials a trunk access code and then presses the Release key or another loop key, the station on the source side and the trunk on the destination side are connected and released from the console. The source can then dial the remaining digits to access an outside destination. The Hold key is ignored.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Activate Recovery on Misoperation of Attendant Console.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
OPT	(AHD) AHA (REA) RED	Autohold on loop Key (denied) allowed. Release on Exclusion (allowed) denied.

Feature operation

This section describes how the feature works in each of the following cases:

- Misoperation of Release key and loop keys
- Misoperation of Autohold on loop key
- Misoperation of Release Source key and Release Destination key.

Misoperation of Release key and loop keys

In the following cases, pressing the Release key or the loop key is ignored:

- Extending a call to a vacant number
- Extending a call to restricted station or trunk
- Extending to a station restricted by Trunk Barring

  Note: Intercept treatment is returned for the above conditions.

- Extending to a partially-dialed number
- Extending a network-blocked call
- Extending a station in the Do Not Disturb mode
- Extending to a station in the Make Set Busy mode
- Extending to a station in the Maintenance-busy state
- Extending to a station in the Line Lockout state
- Extending to a busy extension without Camp-on or Call Waiting
- Extending to a station restricted by Trunk-to-Trunk Connection Restriction
- Releasing from a conference connection – The attendant is prevented from releasing a conference connection, established on the source side, by pressing the Release key or a loop key in the following cases:

  — if there is no destination. Pressing either the Release key or a loop key places the active loop on hold rather than releasing it. The conference can be released by pressing the Release Source key.

— if the attempt to extend the call to the destination was not successful. The conference can be released by pressing the Release Destination key.

— if there is another party already connected as a destination. Pressing the Hold key, Release key or another loop key puts the active loop on hold, rather than releasing it. The destination side can be released by pressing the Release Destination key. The source side can be released by pressing the Release Source key. If an established conference connection cannot be released due to Trunk-to-Trunk Connection Restriction, pressing the Release Source key causes the conference to be released from the console and the trunks disconnected.

Note: Busy tone or overflow tone is returned for the above conditions.

Misoperation of Autohold on the loop key

On a console that is equipped with the Autohold on loop key option, if the attendant is on a call that has terminated properly and presses the loop key while switching to another call, the active loop is placed on hold rather than being released. Besides preventing the inadvertent release of the caller, this option allows the attendant to toggle between any number of held calls by having to press only one key. If the attendant is on a call that cannot be terminated properly, pressing another loop key releases the destination side and puts the source side on hold.

In the following cases, pressing the Release key or the loop key places the call on hold rather than releasing it.

- Extending to a busy extension without Camp-on or Call Waiting, or
- Extending to a station restricted by Trunk-to-Trunk Connection Restriction.

Misoperation of the Release Source/Release Destination key

This option allows the system to ignore the pressing of the Release Source or Release Destination key, preventing the release of either the excluded source or destination party, or a conference call connection. The source or destination party involved in a talking connection with the attendant may still be released by pressing the Release Source or Release Destination key, as appropriate. In a lockout situation, where both source and destination parties are excluded, the attendant may use either the Release Source or Release Destination key to disconnect both parties, since the attendant is not able to re-enter the connection.

Reference Clock Switching

Contents

This section contains information on the following topics:

Feature description 561
Operating parameters 563
Feature interactions 563
Feature packaging 563
Feature implementation 563
Feature operation 564

Feature description

This product improvement allows a Clock Controller reference to automatically switch to another tracking reference if the reference goes into a non-acceptable state (the Clock Controller can track on its primary reference, secondary reference, or be in free run). A non-acceptable state is considered as one of the following:

- The reference loop is disabled.
- For 2.0 Mbps Primary Rate Interface (PRI2), one of the following group 2 errors is detected on the reference loop:
  - The far end is in out-of-service state
  - The far end has lost Multiframe Alignment Signal
  - Alarm Indication Signal is sent

  - — Loss of Frame Alignment, and
  - — Loss of Multiframe Alignment.
- For DTI2, if the reference loop is in Out-of-service (OOS) grade of service, or if the reference loop is in No New Call state, if the OOS is inhibited.

Clock references are supplied to the Clock Controller by the DTI2/PRI2 pack during tracking mode. As mentioned, the Clock Controller can track on its primary reference, secondary reference, or be in free run. If tracking on primary reference and a non-acceptable state is reached, the Clock Controller switches off primary reference and tracks on secondary reference, if it is in an acceptable state, or goes into free run. While tracking in secondary reference, the Clock Controller makes regular periodic checks, at the Clock Controller Audit Rate (CCAR), to determine whether tracking can resume on the primary reference. When the primary reference returns into acceptable state, tracking on primary reference resumes during the next Clock Controller audit.

The same processing occurs if the Clock Controller is tracking on secondary reference, and it goes into a non-acceptable state. It goes into primary reference, if in acceptable state, or free run.

When tracking in free run and an non-acceptable state is encountered, the Clock Controller will first attempt to track on primary state, if in an acceptable state, and then on secondary state. The free run tracking is controlled by a free run guard timer, which is started as soon as tracking begins in free run. As soon as this timer runs out, tracking is attempted on the primary reference and then on the secondary reference. If both are still in a non-acceptable state, tracking continues in free run and the free run guard timer is restarted. If the free run guard timer is not configured, the attempt to switch over to primary or secondary reference is made only as part of the Clock Controller check for an acceptable state on the primary and secondary references.

When the Clock Controller switches from one reference to another, a small delay occurs due to the loop status update and the switching process. During this delay, the reference is given by the Clock Controller to itself in hardware free run state.

Operating parameters

Clock Controller cards QPC775 and NTRB53, and circuit packs QPC915 and QPC536 (2.0 Mbps Digital Trunk Interface), and/or NT8D72AA (PRI2).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Reference Clock Switching requires the following packages:

- International Supplementary Features (SUPP) package 131
- 1.5 Mbps Digital Trunk Interface (PBXI) package 75
- one or both of 2.0 Mbps Digital Trunk Interface (DTI2) package 129 and 2.0 Mbps Primary Rate Interface (PRI2) package 154

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 60 – Enable automatic switch over of system clock sources on the Clock Controller.

2 LD 73 – Enable fast clock switching.

LD 60 – Enable automatic switch over of system clock sources on the Clock Controller.

Command	Description
...	
EREF	Enable automatic switch over of system clocks. Enable automatic switch over of primary and secondary reference clocks. Also enables recovery of primary or secondary clocks when loops associated with these clocks are automatically enabled.

LD 73 – Enable fast clock switching.

Prompt	Response	Description
...		
CCGD	0-(15)-1440	Clock Controller free run Guard time (in minutes).
CCAR	0-(15)	Clock Controller Audit Rate. The time, in minutes, between normal CC audits. Only programmable on units equipped with 2.0 Mbps DTI/PRI.
EFCS	(NO) YES	Enable Fast Clock Switching.

Feature operation

No specific operating procedures are required to use this feature.

Remote Call Forward

Contents

This section contains information on the following topics:

Feature description 565
Operating parameters 565
Feature interactions 566
Feature packaging 568
Feature implementation 568
Feature operation 571

Feature description

Remote Call Forward (RCFW) allows a telephone user to program Call Forward from a remote telephone. With Remote Call Forward (RCFW) enabled, forwarding DNs can be defined and Call Forward All Calls can be activated from within the system or outside the local switch. The Remote Call Forward (RCFW) feature is password protected.

The Station Control Password (SCPW) is required to program Remote Call Forward. Entering a password length of 0 disables the password control for both Electronic Lock and RCFW.

Operating parameters

RCFW requires the following:

- set the password length in LD 15, at the SCPL prompt

- add passwords in LD 10 and LD 11, at the SCPW prompt
- allow Call Forward All Calls in LD 10 and LD 11, and
- define Remote Call Forward Activate (RCFA), Deactivate (RCFD), and Verify (RCFV) Flexible Feature Codes (FFC) in LD 57.

To activate RCFW from outside of the local switch, you must use the Direct Inward System Access (DISA) DN. The telephone's Prime DN is associated with the RCFW password for added security. Also, RCFW can activate or deactivate Call Forward on a telephone, and verify the same feature on a telephone.

Changes to the Station Control Password length do not take effect until after a data dump and SYSLOAD.

If there are two telephones with the same Prime DN, it is recommended that only one of them have a Station Control Password. With RCFW, it is possible that two telephones could have the same password assigned. With the same password, they could control each other's security. For the same reason, the Secondary DN for an Automatic Call Distribution (ACD) telephone should not appear as a Prime DN on another telephone.

A unique number code must be programmed for each of the FFC functions relating to RCFW: Remote Call Forward Activate (RCFA), Remote Call Forward Deactivate (RCFD), and Remote Call Forward Verify (RCFV). You can change the RCFW Directory Number (DN) from your own telephone or from a telephone remote from the switch.

RCFW is not supported for ACD telephones.

Feature interactions

Attendant Administration

Attendant Administration does not support the telephone programming associated with Remote Call Forward.

Call Forward Destination Deactivation

Remote Call Forward (RCFW) and Call Forward Destination Deactivation (CFDD) provide the same functionality but are activated differently. CFDD does not require the call forward station's control password to deactivate the call forward functionality on the call forward station.

The call forwarded destination can use the Remote Call Forward deactivation FFC as well as CFDD to deactivate the Call Forward All Calls functionality on the call forward station.

Call Forward, Internal Calls

Remote CFW Activate (RCFA), Remote CFW Deactivate (RCFD), and Remote CFW Verify (RCFV) FFCs can be used only to access CFW All Calls; they cannot be used to access Internal Call Forward.

China – Flexible Feature Codes - Outgoing Call Barring Enhanced Flexible Feature Codes - Outgoing Call Barring

Activation of CFW to a barred DN by Remote Call Forward will be permitted, since the user has had to dial the Station Control Password, which could also have been used to deactivate Outgoing Call Barring (OCB).

Multiple Appearance Directory Number

With a Multiple Appearance Directory Number (DN) and both sets having a Station Control Password (SCPW), Remote Call Forward may not operate as intended (that is, if Call Forward has been activated using the Remote Call Forward feature, Call Forward remains activated when an attempt to deactivate it is made from the set on which it is active).

Phantom Terminal Numbers (TNs)

If Remote Call Forward is to be used in conjunction with a Phantom TN, the Phantom TNs must be configured with the Call Forward All Calls (CFW) feature.

Preventing Reciprocal Call Forward

This modification applies to Remote Call Forward.

Set-Based Administration Enhancements

A set may be remote call forwarded while someone is actively logged into it with Set-Based Administration login.

2500 Telephone Features

When Flexible Feature Codes (FFC) package 139 is defined and active on your system, a telephone provisioned for Call Forward in LD 10 can also Call Forward All Calls from a remote internal DN.

Feature packaging

The following software packages are required to implement Remote Call Forward:

- Optional Features (OPTF) package 1
- Flexible Feature Codes (FFC) package 139, and
- Controlled Class of Service (CCOS) package 81.

In addition, the following software packages are required to implement RCFW on analog (500/2500 type) telephones:

- Special Service for 2500 (SS25) package 18, and
- 500 Set Dial Access to Features (SS5) package 73.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Set the Station Control Password length.

2 LD 57 – Define Remote Call Forward Flexible Feature Codes.

3 LD 10 – Set the Station Control Password for analog (500/2500 type) telephones and allow Call Forward.

4 LD 11 – Set the Station Control Password for Meridian 1 proprietary telephones and allow Call Forward.

LD 15 – Set the Station Control Password length.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FFC	FFC gate opener.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- SCPL	0-8	Station control password length (0-8). Entering 0 disables the Remote Call Forward and the Electronic Lock features. ***Note:*** A data dump and SYSLOAD are required to implement a change in password length. Shorter passwords are filled with leading zeros. Passwords that are too long have the leading digits truncated.
- FFCS	YES	Change end of dialing digits in FFC.
- - STRL	1-3	Number of digits to indicate FFC end of a feature activation.
- - STRG	(#), xxx	1 to 3 digits to indicate FFC end of a feature entry.

LD 57 – Define Remote Call Forward Flexible Feature Codes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	FFC	Flexible Feature Codes.
FFCT	(NO) YES	FFC Confirmation Tone (optional).
CODE	RCFA	Remote Call Forward Activate.
RCFA	xx	RCFA code
CODE	RCFD	Remote Call Forward Deactivate.

RCFD	xx	RCFD code.
CODE	RCFV	Remote Call Forward Verify.
RCFV	xx	RCFV code.

LD 10 – Set the Station Control Password for analog (500/2500 type) telephones and allow Call Forward.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCPW	xxx...x	Station control password (0-8 digits as defined by prompt SCPL in LD 15).
	X	Entering X deletes the password.
FTR	CFW 4-(16)-23	Allow Call Forward and set forwarding DN length.

LD 11 – Set the Station Control Password for Meridian 1 proprietary telephones and allow Call Forward.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

SCPW	xxx...x	Station control password (0-8 digits as defined by prompt SCPL in LD 15).
	X	Entering X deletes the password.
KEY	xx CFW 4-(16)-23	Assign Call Forward key (xx) and set forwarding DN length.

Feature operation

From any telephone within the system, simply lift the handset and use the following procedures. From any telephone outside the system, first dial the Direct Inward System Access (DISA) number for your system, wait for dial tone, and dial any required passwords and Authorization Codes.

1 Dial the Remote Call Forward Activate FFC.

2 Dial the Station Control Password for the telephone to be forwarded.

3 Dial the Prime DN of the telephone to be forwarded.

4 Dial the number to which calls will be forwarded.

5 Dial the end-of-entry digit(s) (defined in LD 15), if these digits plus the number of digits in the forwarding DN are less than 24 digits. (If you do not dial the end-of-entry digits, the forwarding DN is saved but cannot be verified remotely.)

You will hear a confirmation tone after entering the main extension number, telling you that the password and extension match. You will hear a second special tone after dialing the end-of-entry digits, telling you that the procedure was successful. If you hear a fast busy signal, hang up and try again.

When entering the forwarding DN, you cannot enter more than 23 digits, including the end-of-entry digits. If you attempt to enter a 24th digit, you will hear an overflow tone.

If the forwarding DN plus the end-of-entry digits are not less than 24 digits, hang up after dialing the forwarding DN. The DN is saved but cannot be verified remotely.

To cancel Remote Call Forward:

- Dial the Remote Call Forward Deactivate FFC.

- Dial the Station Control Password for the telephone.
- Dial the Prime DN of the telephone.

To verify Remote Call Forward:

- Dial the Remote Call Forward Verify FFC.
- Dial the Station Control Password for the telephone.
- Dial the Prime DN of the telephone.
- Dial the number to which calls should be forwarded.
- Dial the end-of-entry digit(s).

If the number to which the telephone is forwarding calls does not match your entry in step 4, you will hear a fast busy signal. If the numbers do match, you will hear a confirmation tone after entering the forwarding number, provided the confirmation tone is enabled in LD 57.

When entering the forwarding DN, you cannot enter more than 23 digits, including the end-of-entry digits. If you attempt to enter a 24th digit, you will hear an overflow tone. You cannot use Remote Call Forward Verify for a forwarding DN that was entered without the end-of-entry digits because of too many digits.

Remote Radio Paging

Contents

This section contains information on the following topics:

Feature description 573
Operating parameters 575
Feature interactions 575
Feature packaging 575
Feature implementation 575
Feature operation 576

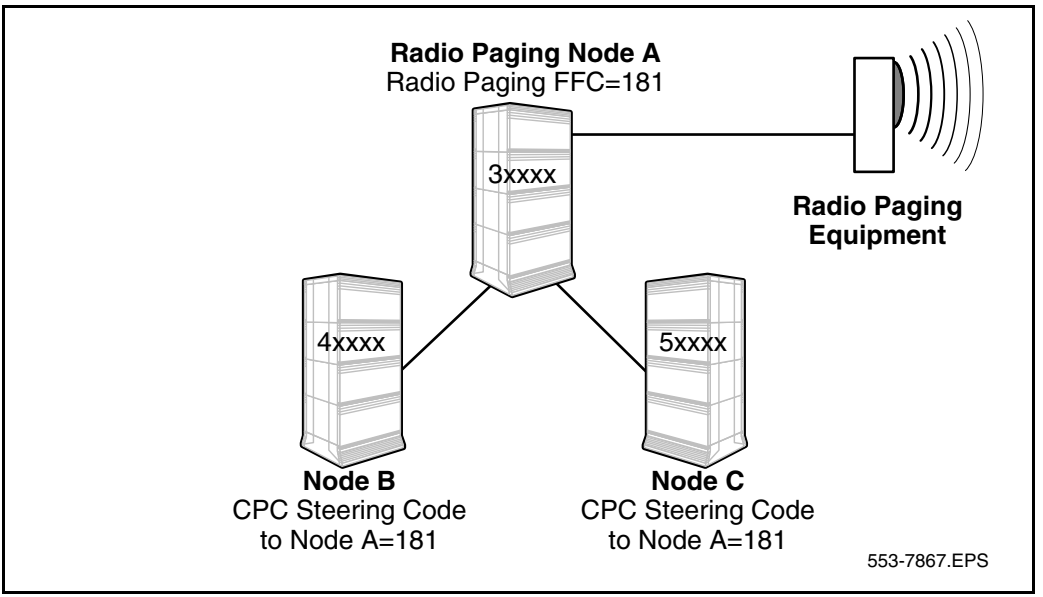
Feature description

This feature provides a network-wide meet-me paging capability from a centralized location. Radio Paging can be accessed by remote nodes through a Coordinated Dialing Plan; however, the Radio Paging feature is not required at remote nodes unless post-selection Radio Paging is required. These remote nodes can define CDP steering codes that route calls to the radio paging node.

These steering codes are the equivalent of Flexible Feature Codes for Radio Paging, and are referred to as *Remote Radio Paging FFCs*. The steering codes must not be deleted by digit manipulation, since the digits are interpreted as the Radio Paging FFC at the Radio Paging node.

Figure 10 demonstrates a possible Remote Radio Paging configuration.

Figure 10
A typical Remote Radio Paging configuration



Node A, which is equipped with the Remote Radio Paging feature, is referred to as the Radio Paging node. The Radio Paging FFC is defined as 181. At remote nodes B and C, steering codes of 181 have been defined to route calls to node A. To access Radio Paging from nodes B and C, a caller simply has to dial 181.

Post Selection Access to Remote Radio Paging

This feature allows the post selection operation of Radio Paging from all nodes in the network. For this functionality, all nodes must be equipped with the Remote Radio Paging feature. For post-selection access, Trunk Steering Codes (TSCs) and Distant Steering Codes (DSCs) are defined as Remote Radio Paging FFCs.

If a post-selection access is made to a set on the same node, the originally-called set must be either ringing or busy. If the originally-dialed set is on another node, it must be on an established call. In this latter case, the established call is disconnected before being routed to the radio paging node.

Post-selection access can be performed from 500/2500-type sets, Meridian 1000 series sets, Meridian proprietary sets, and Attendant Consoles.

Operating parameters

All DNs in the network must have the same fixed length.

The * and # symbols cannot be used as part of the Radio Paging FFC.

Post Selection Access cannot be done using the single-digit access method.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Controlled Class of Service (CCOS) package 81; Flexible Feature Codes (FFC) package 139; and Radio Paging (RPA) package 187.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 87 – Create the Coordinated Dialing Plan TSCs and DSCs for remote nodes.

2 LD 11 – Assign the TSC or DSC steering code to the Radio Paging key on Meridian 1 proprietary telephones.

3 LD 12 – Assign the TSC or DSC steering code to the Radio Paging key on Attendant Consoles.

LD 87 – Create the Coordinated Dialing Plan TSCs and DSCs for remote nodes.

Prompt	Response	Description
...		
DSC	xxxx	Distant Steering Code. Respond with a four-digit value. The DSC must be identical to the Radio Paging FFC at the radio paging node.
- RRPA	(NO) YES	(Disable) enable Remote Radio Paging Access. Remote Radio Paging FFC is being used. Prompted if a CDP, TSC, or DSC is being changed.
TSC	xxxx	Trunk Steering code. Respond with a four-digit value. The TSC must be identical to the Radio Paging FFC at the radio paging node.

LD 11 – Assign the TSC or DSC steering code to the Radio Paging key on Meridian 1 proprietary telephones.

Prompt	Response	Description
...		
KEY	xx RPAG yyyy	Key number, Radio Paging, Route Access Code.

LD 12 – Assign the TSC or DSC steering code to the Radio Paging key on Attendant Consoles.

Prompt	Response	Description
...		
KEY	xx RPAG yyyy	Key number, Radio Paging, Route Access Code.

Feature operation

No specific operating procedures are required to use this feature.

Restricted Call Transfer

Contents

This section contains information on the following topics:

Feature description 577
Operating parameters 577
Feature interactions 577
Feature packaging 578
Feature implementation 578
Feature operation 578

Feature description

This feature provides the Call Transfer Restricted (XFR) Class of Service for analog (500/2500 type) telephones. By assigning XFR Class of Service in LD 10, a Call Transfer attempt will not result in action. This is different from the Call Transfer Denied (XFD) Class of Service, which will route the call to the attendant when a transfer is attempted.

Operating parameters

The Three-party Service Allowed Class of Service, part of the Multiple-Party Operation feature, cannot be used together with the XFR Class of Service.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 10 – Enable Restricted Call Transfer for an analog (500/2500 type) telephone.

Prompt	Response	Description
...		
CLS	XFR	Restrict call transfers and do not recall to attendant.

Feature operation

With XFR Class of Service assigned, a Call Transfer request will not result in action.

Restricted Direct Inward Dialing Class of Service

Contents

This section contains information on the following topics:

Feature description 579
Operating parameters 579
Feature interactions 580
Feature packaging 580
Feature implementation 580
Feature operation 580

Feature description

In order to restrict certain stations from receiving Direct Inward Dialing (DID) calls, the feature will either restrict DID (RDI) calls or unrestricted DID (UDI) calls. The RDI stations will fully restrict DID calls and whereas non-DID calls will be treated according to their normal Class of Service.

Operating parameters

The Central Office must be equipped to handle the special signaling requirements associated with the Restricted DID Class of Service feature described above.

The Restricted DID Class of Service feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Attendant Administration of the Restricted DID Class of Service is not available.

Feature interactions

Class of Service Restrictions

The Restricted DID Class of Service feature changes the access restrictions for telephone sets which have the feature enabled. These sets are treated as fully-restricted with respect to direct calls from DID trunks.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 10 – Enable Restricted Direct Inward Dialing for an analog (500/2500 type) telephones.

Prompt	Response	Description
...		
CLS	(UDI) RDI	This station (is not) is restricted from receiving direct DID calls.

Feature operation

No specific operating procedures are required to use this feature.

Reverse Dial on Routes and Telephones

Contents

This section contains information on the following topics:

Feature description 581
Operating parameters 582
Feature interactions 582
Feature packaging 582
Feature implementation 582
Feature operation 582

Feature description

This feature is used to allow a customer to define their dialpulse format as one of the following:

- regular dial format
- reverse dial format, or
- N+1 dial format.

The feature can be allowed or disallowed on either a route or on all telephones, on a customer basis, by associating a tone table with the route or customer, and setting the reverse dial format in the tone table as required.

Both the “*” and “#” are handled in the same manner as it exists in the regular format. Regular dial format is the default for the feature.

Operating parameters

The feature is supported for Central Office (CO), Foreign Exchange (FEX), Wide Area Telephone Service (WATS), TIE, and Direct Inward Dialing (DID) routes only. Internal system calls are unaffected, except when the feature applies to customers.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Flexible Tones and Cadences (FTC) package 125.

Feature implementation

LD 56 – Configure customer's tone and ringing parameters.

Prompt	Response	Comment
...		
RDVL	(0) 1 2	No Reverse Dial format. Reverse Dial format 1 selected. Reverse Dial format 2 selected.

Feature operation

No specific operating procedures are required to use this feature.

Ring Again

Contents

This section contains information on the following topics:

Feature description 583
Operating parameters 583
Feature interactions 584
Feature packaging 587
Feature implementation 587
Feature operation 588

Feature description

Ring Again gives you the opportunity, after encountering a busy Directory Number (DN), to ring the DN again when it becomes free. If a dialed DN is busy, or if all the trunks are busy, pressing the Ring Again key asks the system to monitor the dialed DN or trunk. When it becomes available, the system notifies you. The call is automatically dialed again when you press the Ring Again key a second time.

When the system alerts you to Ring Again, you have a limited amount of time to respond. Analog (500/2500 type) telephones have six seconds, while Meridian 1 proprietary telephones have 30 seconds.

Operating parameters

A key/lamp pair must be assigned to Meridian 1 proprietary telephones for Ring Again. M3000 and M2317 telephones access Ring Again with a softkey.

Several people can activate Ring Again against the same DN while it is busy. When the DN becomes free, the system notifies the first person in line.

For analog (500/2500 type) telephones, a Special Prefix (SPRE) or Flexible Feature Code (FFC) may be used.

Feature interactions

Attendant Blocking of Directory Number

It is possible to activate Ring Again towards a DN that is blocked due to the Attendant Blocking of DN feature.

Attendant Overflow Position

If Ring Again is activated against the Attendant Overflow Position (AOP) DN, notification is given to the originator when the telephone becomes idle. An AOP call, however, takes precedence over Ring Again notification on the AOP DN when the AOP DN becomes free.

Automatic Set Relocation

If Ring Again is active when a telephone is relocated, the feature is deactivated.

Basic/Network Alternate Route Selection (BARS/NARS)

If the system is equipped with BARS or NARS, the Ring Again feature is used with the Call Back Queuing option to queue for outgoing trunks.

Call Forward/Hunt Override Via Flexible Feature Code

Using the Ring Again feature is possible after using the Call Forward/Hunt Override FFC and encountering a busy signal. Ring Again can be placed against the set for which the Call Forward/Hunt Override FFC was used (that is, the set with CFW active should be rung by the Ring Again feature).

Call Waiting

The user is notified that a previously busy line is free only when both the original call and the waiting call have disconnected.

Calling Party Privacy

A call automatically redialed by the Ring Again – Busy Trunk feature will respect the Calling Party Privacy requested when the call was originally dialed.

Charge Account and Calling Party Number

When Ring Again is activated, no charge record is generated, but the information is stored for future use. If Ring Again is canceled before a trunk is seized, the charge number is deleted and no record is produced. If a trunk is seized later by Ring Again, the charge record is generated in the usual manner. The use of Ring Again with Charge Account ties up system resources because an auxiliary call register must be maintained in the Ring Again queue.

China – Flexible Feature Codes - Outgoing Call Barring
Enhanced Flexible Feature Codes - Outgoing Call Barring

Ring Again cannot be activated after a call is barred by Outgoing Call Barring. Sets with display will not offer Ring Again.

Conference

This feature cannot be activated during a conference call.

Dial Access to Group Calls
Group Call

Ring Again cannot be applied to a Group Call.

Enhanced Override

Ring Again is the only other feature currently available once a busy telephone has been encountered. Ring Again is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Group Hunt

Ring Again will not be supported.

Idle Extension Notification

During the time that an extension is supervised or temporarily blocked from receiving calls due to the Idle Extension Notification feature, it is possible to activate Ring Again towards that extension. It is also possible to request for Idle Extension Notification on an extension that is supervised for Ring Again. When the extension becomes idle, the Idle Extension Notification will be served first.

ISDN QSIG/EuroISDN Call Completion

Analog (500/2500 type) sets can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary sets can make Ring Again requests based on the number of Ring Again keys programmed on a set.

Multi-Party Operations

When a TSA Class of Service analog (500/2500 type) telephone with a call on hold encounters Busy Tone, Ring Again is not possible.

Ring Again is not allowed if the user of an analog (500/2500 type) telephone has a call on hold and receives a busy signal when calling a second party.

Network Intercom

Hot Line calls terminating on a busy key become normal calls. Hence, they may use the Ring Again feature under normal circumstances.

On Hold on Loudspeaker

Ring Again can be applied to a busy loudspeaker DN.

Override
Override, Enhanced

Priority Override

Ring Again is the only other feature currently available once a busy telephone has been encountered. Ring Again is not allowed on an analog (500/2500 type) telephone making a Multi-Party Operations consultation call.

Preference Trunk Usage

Searching for an available trunk via the Ring Again feature is subject to the Preference Trunk Usage feature at trunk seizure time. Earlier trunk availability checks are not carried out.

Feature packaging

Ring Again is included in Optional Features (OPTF) package 1 and has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable Ring Again for analog (500/2500 type) telephones.

2 LD 11 – Enable Ring Again for Meridian 1 proprietary telephones.

LD 10 – Enable Ring Again for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XRD) XRA	Ring Again is (denied) or allowed.

LD 11 – Enable Ring Again for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.

TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx RGA	Ring Again key, where: xx = key number (must be key 27 for M2317 or M3000).

Feature operation

Ring Again is slightly different for each telephone type. Be sure to follow the correct operating instructions.

Meridian 1 proprietary telephones

To activate Ring Again after hearing a busy signal:

- Press **Ring Again**.
- Hang up, or press **Rls**.
- When you hear the Ring Again tone, lift the handset or select a free **DN**.
- Press **Ring Again**. The number is automatically dialed.

To cancel Ring Again:

- Press **Ring Again** before you hear the notification tone.

M3000 Touchphone

To activate Ring Again after hearing a busy signal:

- Press **Ring Again**.
- Hang up, or press **Rls**.
- When you hear the Ring Again tone, lift the handset or select a free **DN**.
- Touch **Connect**. The number is automatically dialed.

To cancel Ring Again:

- Press **Ring Again** before you hear the notification tone.

M2317 telephone

To activate Ring Again after hearing a busy signal:

- Press **RINGAGN**.
- Hang up, or press **Rls**.
- When you hear the Ring Again tone, lift the handset or select a free **DN**.
- Press **Call** . The number is automatically dialed.

To cancel Ring Again:

- Press **Call** before you hear the notification tone.

Analog (500/2500 type) telephones

To activate Ring Again after hearing a busy signal:

- Flash the switchhook or press **LINK**.
- Dial SPRE+1, or the Flexible Feature Code (FFC) assigned.
- When you hear the Ring Again tone bursts, lift the handset while you still hear the ringing. The number is automatically dialed.

To cancel Ring Again:

- Before you hear the notification tone, lift the handset and dial SPRE +2, or the FFC assigned, and hang up.

Ring Again on No Answer

Contents

This section contains information on the following topics:

Feature description 591
Operating parameters 592
Feature interactions 593
Feature packaging 595
Feature implementation 595
Feature operation 597

Feature description

The Ring Again No Answer (RANA) feature extends the capabilities of Ring Again for standalone applications, and Network Ring Again for Integrated Services Digital Network (ISDN) applications. The feature allows Ring Again to be applied to a station that does not answer.

This feature applies to Meridian 1 proprietary telephones, as well as analog (500/2500 type) telephones.

Users of Meridian 1 proprietary telephones, upon encountering a station that does not answer, can activate RANA by pressing the Ring Again (RGA) key. When the desired station goes off-hook, to make or receive a call, and then goes on-hook, the station that activated RGA receives a buzz through the telephone's loudspeaker (while the RGA lamp flashes, if that station is idle). The station user can dial the desired station by lifting the handset or pressing a DN key, and then pressing the RGA key.

Users of analog (500/2500 type) telephones, upon encountering a station that does not answer, can activate RANA by performing a recall, and then dialing the Ring Again Activate Flexible Feature Code, or dialing SPRE then the digit 1. After receiving confirmation dial tone, the user goes on-hook to make or receive calls as usual. When the desired station goes off-hook, to make or receive a call, and then goes on-hook, the station that activated RGA receives six ring cycles as a Ring Again notification (if the station is idle). To dial the desired party, the station user has to go off-hook before the six-ring cycle ends. If the desired party goes off-hook while RANA is being applied, Ring Again Busy is activated instead of RANA.

To deactivate RANA from an analog (500/2500 type) telephone, the user goes off-hook and dials the Deactivate Ring Again or the Deactivate Feature Flexible Feature Code, or dials SPRE then the digit 2.

This feature is described more fully in the *Meridian Link ISDN/AP General Guide* (553-2901-100).

Operating parameters

Ring Again on No Answer cannot be applied:

- if the dialed DN is a Pilot DN
- to Attendant Consoles
- to a station which has been intercepted to the attendant
- to a station which is queued for an attendant
- to a station which has been recalled to an attendant due to misoperation
- to Automatic Call Distribution (ACD) stations
- to a station with Radio Paging active
- to trunks

Meridian 1 proprietary telephones must be equipped with a Ring Again (RGA) key/lamp combination.

Ring Again on No Answer is applied to the originally dialed DN only.

Feature interactions

Attendant Recall

A set that is recalling the attendant cannot apply Ring Again on No Answer.

Call Forward All Calls
Call Forward No Answer

If an unanswered call is forwarded to another station by any of these features, RANA is applied to the originally dialed station.

Call Forward/Hunt Override Via Flexible Feature Code

Using the Ring Again No Answer feature is possible after using the Call Forward/Hunt Override FFC and encountering an idle set that does not answer. Ring Again No Answer can be placed against the set for which the Call Forward/Hunt Override FFC was used (that is, the set should be rung by the Ring Again No Answer feature).

Group Hunting

RANA cannot be applied if the DN dialed was a Pilot DN.

Hunting

If RANA has been applied to a station going through a Hunt sequence, Ring Again is applied to that station and not the ringing station.

Intercept Treatment

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

Intercept to Attendant

RANA cannot be applied by a set that is intercepted to the attendant.

ISDN QSIG/EuroISDN Call Completion

Analog (500/2500 type) sets can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary sets can make Ring Again requests based on the number of Ring Again keys programmed on a set.

Multiple Appearance Directory Number

The Ring Again on No Answer feature will only function on Multiple Appearance Directory Numbers that have been assigned to two different sets provided that both users, with the Ring Again on No Answer activated, go off-hook to make a call and then go on-hook. If both users do not go off-hook then the originator will not receive a buzz through the loudspeaker.

Network Intercom

If Ring Again No Answer is activated for a Hot Type I call, it is activated as if the call had been dialed normally.

Phantom Terminal Numbers (TNs)

Although RANA can be applied to a phantom DN, it is not recommended. Because a phantom DN cannot be active or busy, the caller is not notified when the phantom DN's forward DN does not answer.

Queued Calls

RANA cannot be applied by a set which is being queued for the attendant or is in the attendant queue during Night Service.

Recall to Attendant due to Misoperation

RANA cannot be applied by a set that is recalling the attendant.

Recall to Same Attendant

A telephone that is recalling the attendant cannot apply Ring Again on No Answer.

Multiple Appearance Directory Number

The Ring Again on No Answer feature will only function on Multiple Appearance Directory Numbers that have been assigned to two different sets provided that both users, with the Ring Again on No Answer activated, go off-hook to make a call and then go on-hook. If both users do not go off-hook then the originator will not receive a buzz through the loudspeaker.

Telephones - M2317, M3000, and M3900

For RANA to function on M2317, M3000, and M3900 telephones, the telephones must be configured with a Ring Again (RGA) key. The Ring Again "soft key" will only be displayed when a busy call is encountered and will not be displayed during ring no answer.

Feature packaging

Advanced ISDN Network Services (NTWK) package 148 for network applications.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable the Ring Again on No Answer setting.

2 LD 10 – Enable Ring Again for analog (500/2500 type) telephones.

3 LD 11 – Enable Ring Again keys for Meridian 1 proprietary telephones.

LD 15 – Enable the Ring Again on No Answer setting.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options

...		
OPT	(RND) RNA	Customer options. Ring Again on No Answer (denied) allowed.

LD 10 – Enable Ring Again for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Type of set. Analog (500/2500 type).
...		
CLS	(XRD) XRA	Class of Service options. Ring Again (denied) allowed.

LD 11 – Enable Ring Again keys for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, 3901, 3902, 3903, 3904, 3905.
...		
KEY	RGA	Customer options. Ring Again on No Answer (denied) allowed.

Feature operation

Meridian 1 proprietary telephones

Place and Accept Ring Again on No Answer

Action	Response
1. User A calls user B.	User A receives ringback tone.
2. User A presses the Ring Again (RGA) key.	Indicator associated RGA key turns on steadily.
3. User A either goes on-hook or presses the Release (RLS) key.	Indicator associated with RGA key remains on and user A is now free to receive or make other calls.
4. User B, the user against which Ring Again was placed, goes off-hook to make a call, and then back on-hook.	User A is given a short buzz through the loudspeaker and the indicator associated with the RGA key will begin to flash.
5. User A either picks up the handset or presses a DN key.	User A receives dial tone.
6. User A presses the RGA key.	The user against which the Ring Again was placed is rung and the indicator associated with the RGA key is turned off.

Cancel Ring Again No Answer

Action	Response
1. User A presses the RGA key.	The indicator associated with the RGA goes from flashing to off, and ring again is cancelled.

Analog (500/2500 type) telephones

In the following feature operation description, the term recall refers to performing a register recall which may be performed in a number of different ways. Some examples are:

- Flash the switch hook (that is, the equivalent of hanging up the handset and picking it back up. This on-hook, off-hook is performed in a time period that is less than what the system would consider to be a valid disconnect).
- Press the flash or LINK button if equipped.

Place and Accept Ring Again No Answer

Action	Response
1. User A calls user B.	User A receives ringback tone.
2. User A performs a recall.	User B stops ringing and User A receives special dial tone.
Note: User B must be in a ringing state for more than two seconds before recall is allowed.	
3. User A dials either the Ring Again Activate (RGAA) Flexible Feature Code, or the Special Prefix (SPRE) code followed by the digit "1".	User A receives dial tone indicating that the Ring Again was successfully placed.
4. User A goes on-hook.	User A is now free to receive or make other calls.
5. User B, the user against which the Ring Again was placed, goes off-hook to make a call, and then goes back on-hook.	User A is given six cycles of ringing as notification.
6. If User A picks up the handset before all six ringing cycles are complete.	User B is rung.
7. If user A does not pick up the handset before all six ringing cycles are complete.	Ring Again is cancelled.

Cancel Ring Again No Answer

Action	Response
1. User A goes off-hook.	User A receives dial tone.
2. User A dials either the Ring Again Deactivate (RGAD) Flexible Feature Code, the Deactivate Feature (DEAF) FFC, or the Special Prefix (SPRE) code followed by the digit “2”.	User A receives dial tone indicating that the Ring Again cancellation was successful.

Ring and Hold Lamp Status

Contents

This section contains information on the following topics:

Feature description 601
Operating parameters 602
Feature interactions 602
Feature packaging 602
Feature implementation 603
Feature operation 603

Feature description

The standard lamp-interruption status indication used with the system is 60 impulse per minute (ipm) (flash) for incoming calls and 120 ipm (wink) for held calls on Meridian 1 proprietary telephones, or on terminals emulating Meridian 1 proprietary telephones. This feature, through a Class of Service assigned in LD 11, allows these indicators to be reversed (wink on incoming calls and flash on held calls for all keys that can carry a call, including the group-call key). Data modules with system firmware must use the standard indication of Reverse Lamp Flash Denied Class of Service.

This feature applies to the following key lamps:

- Directory Numbers (DNs)
- Conference
- Transfer

- Voice Call
- Call Waiting
- Dial Intercom Group
- Group Call (For Group Call, a fast blink can be configured to indicate that not all members of a group have answered a group call; a slow flash indicates that a call has been placed on hold by the originator.)
- Automatic Call Distribution (ACD) incalls
- ACD answer agent
- ACD supervisory call, and
- ACD emergency answer.

Operating parameters

This feature cannot be applied to analog (500/2500 type) telephone, M3000 sets, and Attendant Consoles.

This feature is not supported through Attendant Administration.

Feature interactions

Privacy Release

If the Privacy Release feature is activated for multiple-appearance single-call DNs, the blinking rate is based on the Class of Service of each set on which other appearances of the DN occur.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 11 – Modify the data blocks for Meridian 1 proprietary telephones.

Prompt	Response	Description
...		
CLS	(RLFD) RLFA	Reversed Lamp Flash (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Ringback Tone from Meridian 1 Enhancement

Contents

This section contains information on the following topics:

Feature description 605
Operating parameters 605
Feature interactions 606
Feature packaging 606
Feature implementation 606
Feature operation 606

Feature description

With the current ringback handling, some Public Exchange/Central Office (CO) stations do not send the calling party any ringback tone when calling an analog (500/2500 type) telephone. This enhancement provides a calling-party ringback tone, when a call is placed to a system on a 2.0 Mbps digital Central Office (CO) trunk.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 14 – Configure system Ringback Tone.

Prompt	Response	Description
...		
TYPE	COT	Central Office Trunk data block.
...		
CLS	(CORX) CORP	Central Office Ringback (not) provided by the system.

Feature operation

Ringback tone is provided until either the call has been answered by an attendant or abandoned by the originator.

Ringing Change Key

Contents

This section contains information on the following topics:

Feature description 607
Operating parameters 607
Feature interactions 608
Feature packaging 608
Feature implementation 609
Feature operation 609

Feature description

This feature allows the user of an M1000 series or digital telephone to change the ringing/non-ringing designation of a Single Call Ringing (SCR) or Multiple Call Ringing (MCR) directory number (DN) located on one of the telephone's key-lamp strips. This is done by using a Ringing Change (RCK) key.

Operating parameters

This feature does not apply to Private Line DNs.

The ringing designation of the Single Call Non-ringing (SCN) and Multiple Call Non-ringing DN keys cannot be changed by using the RCK key.

This feature requires a separate key/lamp configuration.

Feature interactions

Attendant Blocking of Directory Number

When the SACP key (or Signal Source) key is pressed to ring a blocked SCR where the Ring Change feature is activated, an audible ring signal will always be given. This is independent of the Ring Change status.

Directory Number Delayed Ringing

If an SCR/MCR key is toggled from “ringing” to “non-ringing”, the Directory Number Delayed Ringing (DNDR) feature will apply to the key. If an SCR/MCR key is toggled again from “non-ringing” to “ringing”, the key will be rung immediately and DNDR will no longer apply.

If an SCN/MCN key is toggled from “non-ringing” to “ringing”, the DNDR key will ring immediately and DNDR will no longer apply. If an SCN/MCN is toggled again from “ringing” to “non-ringing”, the key will not ring immediately and the DNDR feature will apply to the key.

Network Intercom

The ringing/non-ringing mode of an enhanced Hot Type D or of a Hot Type I key is not changeable by using the Ringing Change Key feature.

Feature packaging

International Supplementary Features (SUPP) package 131; and Ringing Change Key (RCK) package 193.

Feature implementation

LD 11 – Define a Ringing Change Key (RCK) for each Meridian 1 proprietary telephone to be equipped with one.

Prompt	Response	Description
...		
KEY	xx RCK y z	Key number, Ringing Change Key, first key lamp strip, second key lamp strip controlled by the key. y = (0)-7 z = 0-(3)-7 Only one RCK key per set is permitted.

Feature operation

Pressing the **RCK** key places the telephone in the Make Set Busy state. Incoming calls to the set receive busy tone, and Multiple Appearance DN calls terminate on another telephone. Pressing an idle **SCR or MCR DN** key indicates the ringing status of the key; a lit key lamp indicates a non-ringing status, and a flashing key lamp indicates a ringing status. Pressing the **SCR or MCR DN** key again changes the ringing status of the key. Pressing the **RCK** again stores the change, and causes the SCR or MCR key lamp to go dark.

During a system initialization a telephone is rendered in the Make Set Busy state. If both the Ringing Change Key and Make Set Busy features are equipped on a telephone, and an initialization occurs during operation of the **RCK** key, the RCK lamp goes dark to inform the user that the changes have not been stored. The MSB lamp is lit to inform the user that the telephone is still in Make Set Busy mode.

Ringing instead of Buzzing on Digital Telephones

Contents

This section contains information on the following topics:

Feature description 611
Operating parameters 612
Feature interactions 612
Feature packaging 613
Feature implementation 613
Feature operation 614

Feature description

The Ringing instead of Buzzing feature, allows a digital telephone to ring when a call is presented as follows:

- when the handset is off hook but the telephone is idle
- when the handset is off hook but the telephone is idle and when the user is busy on another line

Ringing alerts a user in a more obvious way than buzzing (previous operation).

If a call is presented to the telephone, it rings according to the Distinctive Ringing Class of Service (DRG1, DRG2, DRG3, and DRG4), instead of buzzing.

There are two Classes of Service which can be assigned in LD 11:

- RNGI (the set rings when idle but off hook and a call is presented)
- RNGB (the set rings when busy or idle, but off hook and a call is presented.

Operating parameters

This feature does not affect the features where a buzz is already provided, such as Ring Again or Manual Signaling.

Buzzing is the default configuration.

Ringing features such as Ringing Change Key, Network Distinctive Ringing or Executive Distinctive Ringing, if implemented, affect the way in which the telephone rings.

Any digital telephone can be assigned an RNGI or RNGB Class of Service.

This feature does not affect Attendant Consoles.

If an attempt is made to enter CLS BUZZ, RNGI or RNGB on an analog telephone programmed in LD 11, a service change error message is output.

Feature interactions

ACD calls

This feature affects calls to an M2216 telephone. Ringing is given to the agent when the CLS is programmed for ringing and the telephone is idle.

Hunting

For telephones with more than one DN, the RNGB Class of Service and Short Hunting programmed calls will ring, not buzz, when the telephone is already busy.

Short Hunting allows calls to hunt to the next higher available key on a proprietary telephone, when a call is already established on a DN key.

Short Buzz for Digital Telephones

The Ringing instead of Buzzing feature takes precedence over the Short Buzz for Digital Telephones feature.

Third Party Applications

Applications which attach to or emulate a digital telephone can be affected by this feature.

Tones, Flexible Incoming

The Ringing instead of Buzzing feature takes precedence over the Tones, Flexible Incoming feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 11 – Configure the Ringing instead of Buzzing feature on digital telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	xxxx	Valid telephone types: 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3901, 3902, 3903, 3904, 3905.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(BUZZ)	Buzz (default).
	RNGI	Ringing applied when telephone is idle but off hook.
	RNGB	Ringing applied when telephone is idle but off hook or busy on the other line.

Feature operation

No specific operating procedures are required to use this feature.

Room Status

Contents

This section contains information on the following topics:

Feature description 615
Operating parameters 617
Feature interactions 618
Feature packaging 619
Feature implementation 620
Feature operation 622

Feature description

Room Status allows customers equipped with a Background Terminal (BGD) to store and retrieve data pertinent to the occupancy, readiness, or cleaning status of any guest room or group of guest rooms.

When equipped with the Room Status software, the system provides the following Room Status information:

- Guest registration and occupancy

  OC (occupied)
  VA (vacant)
  CH (check in)
  CH OU (check out)

- Cleaning status

  RE (cleaning required)
  PR (cleaning in progress)
  CL (room cleaned)
  FA (failed inspection)
  PA (passed inspection)
  SK (cleaning skipped)

- Sale status

  NS (not for sale)
  SA (ready for sale)

- Other status information

  CCOS (Controlled Class of Service)
  DND (Do Not Disturb)
  MW (Message Waiting)
  CA (Category one – 1 to 15)
  TL (telephone check)

Do Not Disturb (DND) has been enhanced for interaction with Room Status on an analog (500/2500 type) telephones. A new customer option allows a visual indication of when the analog (500/2500 type) telephone is in the DND mode: the lamp on the telephone lights up.

The Room Status feature provides four methods of accessing the Room Status data:

- Off-hook detection: Hotel and hospital staff generally clean occupied rooms during certain hours of the day. From a Background Terminal (BGD), an option can be entered to set all occupied rooms to “cleaning status request” mode for a predefined time-of-day interval. During this interval, the system monitors the room telephone’s switchhook state to detect a change in the Room Status.

- Dial Access: This method is an enhancement to the off-hook detection method for updating the room cleaning status. This method offers seven cleaning-status options, as compared to the two offered by off-hook detection. Again, you allow or deny the dial access method by using the Background Terminal commands.
- Room Status key: A Room Status key (RMK) can be provided on an M1109 or Meridian 1 proprietary set. This allows the telephone user to read or alter the status of any room in the system.
- Background Terminal: The Room Status feature is administered from a Background Terminal (BGD) assigned to the customer. BGDs are defined in the configuration record and are connected to the system through a Serial Data Interface (SDI) port. Devices used as BGDs can be any ASCII serial terminal conforming to EIA RS-232C or CCITT V.24 standards.

Operating parameters

The Room Status key (RMK) is supported only on telephones equipped with a display.

A room telephone is defined with Controlled Class of Service allowed (CCSA). The following telephones are supported as room phones:

- Analog (500/2500 type) telephones
- M1309 telephones
- Meridian 1 proprietary sets

The M3000, M2317, and ACD telephones are not supported as room phones. Room Status is not supported on telephones with DTA (data terminal allowed) Class of Service. The RMK is not supported on Attendant Consoles.

A room telephone is allowed to change the status of its own room.

The Room Status feature is mutually exclusive with the Multiple-Tenant, Centralized Attendant Service (CAS), and Coordinated Dialing Plan (CDP) features.

A message center must be defined for the Do Not Disturb (DND) visual indication function on an analog (500/2500 type) telephones. This is mutually exclusive of Integrated Messaging System (IMS) and Meridian Mail Message Centers.

All analog (500/2500 type) telephones that are to use the Do Not Disturb (DND) visual indication must also have an LPA (Lamp Allowed) Class of Service.

Feature interactions

Attendant Administration

Room Status is not supported by Attendant Administration.

Automatic Wake Up

Room Status and Automatic Wake Up both use the Background Terminal (BGD). If the WAKE option is selected for the check-in/check-out operation, the wake-up call for that room is canceled after a check-in or check-out operation.

Automatic Wake Up FFC Delimiter

When a guest has either checked in or out, the room status changes. If an AWU request is still active, it is canceled if it is included as part of the Check In/Out option.

Controlled Class of Service

You can change the access restrictions for room telephones from the BGD or from a telephone equipped with a Room Status key (RMK).

Hot Line

The Room Status feature is incompatible with any telephone for which going off-hook activates Hot Line.

Maid ID

Maid ID is not required but is recommended to track maid performance. The Maid ID must be entered each time the Room Status changes, or it will not be recorded.

Multiple Tenant

Telephones equipped with an RMK can change the Controlled Class of Service (CCOS) of telephones for any tenant in a Customer Group.

Off-Hook Alarm Security

Cleaning changes entered using the Off-Hook Detection Method are mutually exclusive with the Off-Hook Alarm Security (OHAS) feature. OHAS takes precedence over the off-hook detection method of the Room Status feature. If a telephone is defined with the Alarm Security Allowed (ASCA) Class of Service, the off-hook detection method does not work.

Feature packaging

Room Status (RMS) package 100 requires the following:

- Controlled Class of Service (CCOS) package 81, and
- Background Terminal Facility (BGD) package 99.

For lamp status, the requirements are as follows:

- Do Not Disturb, Individual (DNDI) package 9, and
- Message Waiting Center (MWC) package 46.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable Controlled Class of Services (CCOS) for analog (500/2500 type) telephones requiring Room Status updates.

2 LD 11 – Enable Room Status key (RMK) for digit display telephones used for Room Status.

3 LD 15 – Change Customer Data Block to allow (or disallow) visual indication of Do Not Disturb (DND) feature. Offered on the customer level, this applies only to analog (500/2500 type) telephones equipped with a Message Waiting (MW) lamp.

Note: This procedure assumes that a BGD has been assigned. Refer to *Hospitality Features: Description and Operation* (553-3001-353) for a complete description and list of commands for the Background Terminal.

LD 10 – Enable Controlled Class of Services (CCOS) for analog (500/2500 type) telephones requiring Room Status updates.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(CCSD) CCSA	Controlled Class of Service (denied) allowed.

LD 11 – Enable Room Status key (RMK) for digit display telephones used for Room Status.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	ADD DDS	Automatic digit display enabled. Digit display enabled.
KEY	xx RMK	Room Status key.

LD 15 – Change Customer Data Block to allow (or disallow) visual indication of Do Not Disturb (DND) feature. Offered on the customer level, this applies only to analog (500/2500 type) telephones equipped with a Message Waiting (MW) lamp.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- DNDL	YES (NO)	Indicator goes on when DND is active. Indicator does not go on (the default).
...		

TYPE	CCS	Gate opener.
- CCRS	UNR	Unrestricted call service.
	CUN CTD TLD SRE FRE FR1 FR2	With CCOS active, the restrictions entered apply.

Feature operation

To read the Room Status by using the RMK (display needed):

- Without lifting the handset, press the **RMK key**.
- Dial the Directory Number (DN) of the room telephone. The DN is displayed, followed by a dash and a two-digit code.
- The first digit indicates occupancy: zero (0) means vacant, one (1) means occupied.

  The second digit indicates Room Status:

  — 1 = RE (cleaning required)

  — 2 = PR (cleaning in progress)

  — 3 = CL (cleaned)

  — 4 = PA (passed inspection)

  — 5 = FA (failed inspection)

  — 6 = SK (cleaning skipped), and

  — 7 = NS (not for sale).

To change the Room Status by using the RMK:

- Without lifting the handset, press the **RMK key**.
- Dial the Directory Number (DN) of the room telephone.
- Dial the new room status as follows:

  — 1 = RE (cleaning required)

  — 2 = PR (cleaning in progress)

  — 3 = CL (cleaned)

— 4 = PA (passed inspection)

— 5 = FA (failed inspection)

— 6 = SK (cleaning skipped), or

— 7 = NS (not for sale).

- Press the **RMK key**.

To change the Room Status by using Dial Access (from the room telephone):

1 Lift the handset and dial SPRE 86.

2 Dial the room status as shown below:

— 1 = RE (cleaning required)

— 2 = PR (cleaning in progress)

— 3 = CL (cleaned)

— 4 = PA (passed inspection)

— 5 = FA (failed inspection)

— 6 = SK (cleaning skipped), or

— 7 = NS (not for sale).

3 Dial * and the Maid ID followed by #, if required.

4 Hang up or press **Rls**.

Note: For complete details on the Room Status operation, see *Background Terminal User Guide*.

Scheduled Access Restrictions

Contents

This section contains information on the following topics:

Feature description 625
Operating parameters 626
Feature interactions 627
Feature packaging 630
Feature implementation 631
Feature operation 639

Feature description

The Scheduled Access Restrictions (SAR) feature allows a customer to define Trunk Group Access Restrictions (TGAR), Class of Service (COS) restrictions, and Network Class of Service (NCOS) restrictions for different hours and days (typically off-hours and off-days). These TGAR, COS, and NCOS restrictions comprise SAR groups. Each customer may define up to 1000 SAR groups, and one of these groups can be assigned to each customer station or route. Up to eight time periods can be defined for each SAR group, and different restrictions may be applied to each time period.

SAR can be overridden on a single call basis for a station or route by using an authorization code or forced charge account. By using the SAR Disable (SADS), SAR Enable (SAEN), SAR Lock (SALK), or SAR Unlock (SAUN) Flexible Feature Codes these restrictions can be changed on a more permanent basis.

SADS returns the set/route to its normal restriction state. SAEN cancels SADS, returning the set to its SAR state. SALK will occur automatically at a predefined period of time or when the Lock command is dialed by the user. Lock restrictions remain in effect until an SAUN or SADS command is entered. The SALK command can be used on a customer basis or SAR group basis, depending on the authcode used.

Typically, the Flexible Feature Codes can be used to do the following:

- extend off-hour restrictions for weekends or holidays (SALK)
- return to the schedule of access restrictions (SAUN)
- extend normal restrictions into the off-hour period for after hour services (SADS)
- cancel this after hour service (SAEN)
- cause off-hour restrictions to start immediately (SALK followed by SAEN), and
- disallow any calls on an Attendant Console (SALK on SAR group containing the attendant(s)).

Customer attendants that are included in SAR groups are placed in Position Busy when an off-hour or off-day period goes into effect. The restricted attendant can only release existing calls or dial the SAR Flexible Feature Codes. New calls cannot be made. Incoming calls will be directed to any other attendants that are not included in SAR groups and that are not in Position Busy.

If the system is placed in Night Service by an attendant, or the system is automatically placed in Night Service because all attendants are in the Position Busy state, incoming calls are routed to the Night DN. Going into Night Service will automatically place attendants who belong to a SAR group into an SAR Locked and Enabled state. These attendants can only release existing calls or dial the SAR Flexible Feature codes; they cannot make new calls when restricted by SAR.

Operating parameters

The definition of authorization codes for SAR decreases the number of authorization codes available for non-SAR use.

SAR does not apply to Direct Inward System Access (DISA) DNs. DISA can be used to manually modify the SAR schedule using an FFC authorization code.

Telephones and trunks assigned to SAR groups have their Class of Service (COS), Trunk Group Access Restriction (TGAR) and Network Class of Service (NCOS) defined by the SAR schedule of their SAR group.

During the periods that a SAR or SAR lock is in effect, the Controlled Class of Service (CCOS) for the station or trunk is overridden.

If a Facility Restriction Level (FRL) is changed in order to be associated with a different NFCR tree, the NCOS using that FRL is affected. Also, different FRLs, and therefore different New Flexible Code Restriction (NFCR) trees, are used at different times according to the NCOS assigned to the SAR group.

Feature interactions

Access Restrictions

The Trunk Access Restriction Group (TARG) defined for each route is not altered by Scheduled Access Restrictions. Access to the route is denied to any telephone or trunk assigned a Trunk Group Access Restriction code that is part of the TARG.

Automatic Redial

The Scheduled Access Restrictions (SAR) on Automatic Redial (ARDL) redialed calls are set when the call is initiated. If restrictions are changed later, the prior restrictions still apply.

Attendant Clearing during Night Service

Attendant Clearing during Night Service should be equipped with Scheduled Access Restriction (SAR) due to the fact that when Night Service is in effect the only operations that may be performed from Attendant Consoles which are members of a SAR group are:

- release any existing calls, or
- dial the one of the following SAR Flexible Feature Codes:
  - Scheduled Access Disable (SADS)

— Scheduled Access Enable (SAEN)

— Scheduled Access Lock (SALK), or

— Scheduled Access Unlock (SAUN).

Authorization Code Security Enhancement

Authorization Codes can be used to override SAR restrictions. In addition, Authorization Codes are defined for the specific use of SAR FFCs.

Basic Alternate Route Selection

If SAR is equipped when Basic Alternate Route Selection (BARS) is set up, a NCOS value between 0 and 99 must be defined for each time period.

Call Detail Recording

If configured, Call Detail Recording (CDR) A-type records are printed for SAR Flexible Feature Codes functions.

Charge Account, Forced

Forced Charge Account (FCA) can be used to override Scheduled Access Restrictions (SAR) on a per-call basis, provided the current Class of Service (COS) of the telephone or trunk is CUN, TLD, or CTD. The current COS is the COS in force according to the SAR schedule. If an Authorization Code that sets the COS to CUN, TLD, or CTD is dialed before the FCA, the call is allowed. FCA sets the COS to UNR and the Network COS (NCOS) to the NCOS defined in LD 15, provided that FCA is enabled on both a customer and telephone/trunk basis.

Class of Service

Sets defined in LD 10 and 11, and trunks defined in LD 14 which are assigned a SARG number, have their Class of Service defined by the SAR schedule of their SAR group.

Controlled Class of Service

During normal hours, Controlled Class of Service (CCOS) restrictions override normal telephone restrictions. During off-hour periods or times when a Scheduled Access Restrictions (SAR) LOCK is in effect, however, Scheduled Access Restrictions apply. When the LOCK or off-hour period ends, CCOS restrictions continue to apply until they are removed or SAR becomes effective again. Whether a CCOS controller or electronic lock is used to activate CCOS, there is no indication to the user when Scheduled Access Restrictions are in effect, overriding CCOS restrictions. A telephone defined in LD 10 or 11 or a trunk defined in LD 14, which is assigned an SAR group number, has its Class of Service defined by the SAR schedule of its SAR group.

Coordinated Dialing Plan

If SAR is equipped when Coordinated Dialing Plan (CDP) is set up, a NCOS value between 0 and 99 must be defined for each time period.

Direct Inward System Access

Direct Inward System Access (DISA) numbers are not assigned to SAR groups and therefore are not affected by SAR schedules.

DISA can be used to manually modify the SAR schedule, provided that the correct FFC and Authorization Code are dialed.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The SAR feature overrides Electronic Lock.

Multi-Tenant Service

If a SAR is assigned to a tenant, any set belonging to the tenant will follow this SAR schedule unless the set belongs to a SAR group. The set's Scheduled Access Restrictions override any SAR assigned to the tenant.

Network Alternate Route Selection

If SAR is equipped when Network Alternate Route Selection (NARS) is set up, a NCOS value between 0 and 99 must be defined for each time period.

Network Class of Service

When a Network Class of Services (NCOS) is changed, it may be necessary to alter the NCOS values defined for each SAR roup in LD 88. The NCOS value, which defines the facility restriction level and hence the NFCR trees, is used as defined by the SAR schedule. Also, different FRLs, and hence different NFCR trees, are used at different times according to the NCOS assigned to the SARG.

New Flexible Code Restriction

If a Facility Restriction Level (FRL) is changed to be associated with a different NFCR tree, any NCOS which uses that FRL is affected. In turn, the NCOS assigned to a SAR group may also be affected.

Office Data Administration System

Office Data Administration System (ODAS) can be used to indicate that telephones have been assigned to an SAR group. ODAS must be equipped in order to print members of a SAR group in LD 81.

Position Busy with Call on Hold

If an attendant in a Scheduled Access Restriction group has a call on hold, the attendant is not placed in Position Busy when the group enters an off-hour period.

Speed Call
Network Speed Call

The System Speed Call and Network Speed Call features ignore the Class of Service and TGAR access restrictions in a SAR schedule, using the Class of Service and NCOS defined in the speed call list.

Trunk Group Access Restriction

SAR does not alter the Trunk Group Access Restriction defined per route.

Feature packaging

Scheduled Access Restrictions requires the following packages:

- Scheduled Access Restrictions (SAR) package 162

- Call Detail Recording (CDR) package 4 for CDR functionality
- Network Authorization Code (NAUT) package 63
- Multi-Tenant Service (TENS) package 86 for Multi-Tenant functionality
- Flexible Feature Codes (FFC) package 139 and Basic Authorization Codes (BAUT) package 25 for the manual modification of schedules
- Network Class of Service (NCOS) package 32 to make NCOS restrictions effective
- Charge Account for CDR (CHG) package 23, Charge Account/ Authorization Code (CAB) package 24, and Forced Charge Account (FCA) package 52 for additional billing information

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 88 – Configure Scheduled Access Restrictions data block.

2 LD 88 – Configure the Authcode data block not to automatically generate Authcodes.

3 LD 88 – Define SAR entries in the Authcode entries data block.

4 LD 10 – Assign individual annalog(500/2500 type) telephones to the selected SAR group in response to the SGRP prompt.

5 LD 11 – Assign individual proprietary telephones to the selected SAR group in response to the SGRP prompt.

6 LD 12 – Assign individual Attendant Consoles to the selected SAR group in response to the SGRP prompt.

7 LD 16 – Assign individual trunk route to the selected SAR group in response to the SGRP prompt.

8 LD 57 – Define Flexible Feature Codes for the SAR disable, SAR enable, SAR lock, and SAR unlock functions.

9 LD 93 – Assign a SARG for each tenant by responding to the TEN prompt with the tenant number and the SGRP prompt with the number of the SAR group to be assigned to the tenant.

LD 88 – Configure Scheduled Access Restrictions data block.

Prompt	Response	Description
REQ	NEW CHG	Create or change existing data block.
TYPE	SAR	Scheduled Access Restrictions.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15). ***Note:*** Prompt will not appear to a user with an LAO password.
SGRP	0-999	SAR group number.
SCDR	(NO) YES	(Do not) activate CDR for the SAR FFC commands.
OFFP	1-8	Off-hour period number. Off-hour periods may overlap; the period that starts first has priority until that off-hour period is over.
	<CR>	Go to ICR prompt.
- STAR hh mm	hh mm	Start time. The current start time (hours and minutes) is printed individually after the prompt. Respond with the new start time.
	X	Remove value and return to OFFP prompt.
- STOP hh mm	hh mm	Stop time. The current stop time (hours and minutes) is printed individually after the prompt. Respond with the new stop time.
	X	Remove value and return to OFFP prompt.

- DAYS	d ... d	Respond with a new set of days to be used. Maximum of seven entries in the range of 1-7. Day 1 = Sunday, Day 2 = Monday, etc.
- COS		Off-hour period Class of Service.
	(UNR)	Unrestricted
	CTD	Conditionally Toll-Denied
	CUN	Conditionally Unrestricted
	FR1	Fully Restricted Class1
	FR2	Fully Restricted Class 2
	FRE	Fully Restricted
	SRE	Semi-restricted
	TLD	Toll Denied
- TGAR	(0)-15	Trunk Group Access Restriction.
- NCOS	0-99	Network Class of Service.
- ICR	(NO) YES	Incoming Calls are Restricted.
LOCK	(1)-8	The LOCK prompt is used to indicate which off-hour period is to be used as the LOCK period. The default is Period 1.

LD 88 – Print the status of the tenant SAR group.

Prompt	**Response**	**Description**
REQ	PRT	Print.
TYPE	SAR	Scheduled Access Restrictions.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password.
TEN	1-511	Tenant number.
SGRP	0-999	Prompted only if no tenant number is entered.

Note: If the system is in an off-hour or locked period when a print command is issued, an asterisk appears following the restrictions being used.If lock is in effect, an additional asterisk appears following the lock prompt.

LD 88 – Configure the Authcode data block not to automatically generate Authcodes.

Prompt	Response	Description
REQ	NEW	New.
TYPE	AUB	Authcode data block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15).
ALEN	1-14	Number of digits in authcodes.
ACDR	YES NO	Activate CDR for authcodes (there is no default response).
RANR	0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems
CLAS	(0)-115	Classcode value assigned to authcode.
AUTO	NO	Do not automatically generate Authcodes. ***Note:*** Prompted when NAUT package 63 is equipped and REQ = NEW. The Authcode length must be a minimum of four digits.

LD 88 – Define SAR entries in the Authcode entries data block.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	AUT	Authcode entries data block.
CUST	xx	Customer number, as defined in LD 15

SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15).
CODE	xxx..x	Authcode (1-14 digits).
SARC	YES NO	Allow or disallow Authcode to be used as the SAR authorization code.
- SERV		SAR service functions for SARC (the SERV prompt appears if SARC = YES)
	(END) ENA (LKD) LKA (DSD) DSA (UND) UNA	Enable (Denied) Allowed. Lock (Denied) Allowed. Disable (Denied) Allowed. Unlock (Denied) Allowed ***Note:*** Up to four entries can be made at once.
- SRGP	0-999 ALL	Number of SAR group to be defined or changed. Change all SAR groups.
CLAS	(0)-115	Class code value assigned to authcode. Cycle continues with CODE. When type = AUT, enter X to configure the authcode as an exempt code. When this data is printed, the month the authcode was deactivated is output. The default is 0 when adding authcode entries.
	X	Exempt authcode.

LD 10 – Assign individual annalog(500/2500 type) telephones to the selected SAR group in response to the SGRP prompt.

Prompt	**Response**	**Description**
REQ:	New CHG	Add, or Change.
TYPE:	500	Analog(500/2500 type) telephone. See *Software Input/Output: Administration* (553-3001-311).
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 11 – Assign individual proprietary telephones to the selected SAR group in response to the SGRP prompt.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Meridian 1 proprietary telephone. See *Software Input/Output: Administration* (553-3001-311).
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 12 – Assign individual Attendant Consoles to the selected SAR group in response to the SGRP prompt.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	1250 2250 ATT	Attendant Console type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Note: Attendant Consoles do not follow the SAR restrictions defined by the SGRP, but they can be locked by using SAR FFCs

LD 16 – Assign individual trunk route to the selected SAR group in response to the SGRP prompt.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

LD 57 – Define Flexible Feature Codes for the SAR disable, SAR enable, SAR lock, and SAR unlock functions.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	FFC	Flexibly Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
...		
CODE	aaaa	Specific Flexible Feature Code Type. To change a specific Flexible Feature Code, enter the associated mnemonic then carriage return <CR>. The mnemonic will then be prompted and the Flexible Feature Code can be entered. The Flexible Feature Code may be up to four digits or up to seven digits if DNXP package 150 is equipped.
- SADS	xxxx	SAR Disable code.
- SAEN	xxxx	SAR Enable code.

- SALK	xxxx	SAR Lock code.
- SAUN	xxxx	SAR Unlock code.

LD 93 – Assign a SARG for each tenant by responding to the TEN prompt with the tenant number and the SGRP prompt with the number of the SAR group to be assigned to the tenant.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TGEN	Tenant SAR data block.
CUST	xx	Customer number, as defined in LD 15
...		
TEN	1-511	Tenant number.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Feature operation

Modification of SAR Restrictions

SAR restrictions can be modified on a per call basis by using an Authorization code, if the Basic Authorization Code (BAUT) package 25 is equipped.

Also, if the Authorization Code and Flexible Feature Codes packages are equipped, the off-hour periods can be shortened or extended by using the four SAR FFCs.

The Authorization Code feature can be used to allow a user to override a Scheduled Access Restriction on a single-call basis by dialing an Authorization Code (Authcode). Each Authcode is assigned a Class of Service, Trunk Group Access Restriction, and a Network Class of Service. The restrictions associated with the dialed Authcode, apply to the call being made. Thus, by using an Authcode, any facility to which access is allowed depending on the restrictions associated with an Authcode, can be accessed by dialing the set, even though the set may normally be denied access.

Single-Call Modification

The Scheduled Access Restrictions feature does not modify using Authcodes to allow calls to be made on restricted sets. Dial either “SPRE + 6” or the AUTH FFC plus the Authcode associated with the desired restrictions. Once dial tone is returned, indicating a valid code, the call may be dialed as normal.

Off-Hour Period Modification

The SADS, SAEN, SALK, and SAUN FFCs defined in LD 57 can be used to modify off-hour period restrictions, by simply dialing the FFC plus an appropriate Authcode. The Authcode determines if the requested function is allowed and whether the action is to take place on a SAR group or a customer basis. An FFC plus and Authcode for a specific SARG is only accepted from a station within that group, or from a station within a tenant which uses that SAR group.

Entering a Flexible Feature Code plus an Authcode results in the following:

- SALK + Authcode = extend off-hour restrictions for weekends or holidays
- SAUN + Authcode = return to the schedule of access restrictions

- SADS + Authcode = extend normal restrictions into the off-hour period for after hour services
- SAEN + Authcode = cancel this after hour service
- SALK followed by SAEN + Authcode = cause off-hour restrictions to start immediately, and
- SALK on SAR group containing the attendant(s) + Authcode = disallow any calls on an Attendant Console.

Scheduled Electronic Lock

Contents

This section contains information on the following topics:

Feature description 641
Operating parameters 642
Feature interactions 643
Feature packaging 644
Feature implementation 644
Feature operation 648

Feature description

Scheduled Electronic Lock (SELK) enhances the Electronic Lock feature.

The Scheduled Electronic Lock feature automatically locks telephone sets at predetermined times. These times are defined in the Scheduled Access Restrictions (SAR) database (LD 88). SAR group numbers are also defined in LD 88. A maximum of eight scheduled lock times can be assigned to each group.

Each telephone set that requires Scheduled Electronic Lock functionality must be assigned to a SAR group in LD 10 or 11, and must have the Scheduled Electronic Lock Allowed (SLKA) Class of Service assigned.

In order to override the Scheduled Electronic Lock feature, the user must use the existing Electronic Lock feature. The user enters the Electronic Lock Deactivated (ELKD) Flexible Feature Code (FFC). The telephone set remains unlocked until the user dials the Electronic Lock Activated (ELKA) FFC. If the user does not dial the ELKA FFC, the system automatically locks the telephone set at the next scheduled lock time. For the set to be unlocked again, the user must dial the ELKD FFC to unlock the telephone set. The telephone set does not automatically unlock.

A special dial tone, defined in LD 56, notifies the user that the telephone set is in a locked state.

Scheduled Electronic Lock Example

The Scheduled Electronic Lock is scheduled for 18:00, 24:00, 02:00. At 22:00, an employee who is working overtime needs to use their telephone set. That person enters the ELKD FFC on the telephone set to unlock it. At 24:00, the telephone set automatically locks, if it has not already been locked by the user. To use the telephone set again, the employee must unlock it. At 02:00, the next scheduled lock time, the telephone set locks once more. The Scheduled Electronic Lock feature remains in effect until the employee unlocks the set by dialing the ELKD FFC.

Operating parameters

The Scheduled Electronic Lock feature supports analog (500/2500 type) and digital telephones on Remote Office.

The Scheduled Electronic Lock feature does not support ACD sets, trunks or PC Attendant.

If a telephone set does not support the SAR and Electronic Lock features (for example, ACD sets), then it will not support the Scheduled Electronic Lock feature.

If the Class of Service (CLS) is set to Scheduled Electronic Lock Deactivated (SKLD) in LDs 10 and 11, the existing Electronic Lock and SAR feature functionality apply.

When the Scheduled Electronic Lock feature is active, it does not take the Controlled Class of Service (CCOS) restriction from LD 15. Configuration is done in LD 88.

When a telephone set is unlocked, CCOS restrictions (if active) override normal telephone set restrictions. When the Scheduled Electronic Lock feature is active, the Scheduled Access Restrictions override the CCOS restrictions.

If the system is busy the Scheduled Electronic Lock feature could be slightly delayed. In this case, it is possible that a user could still dial an external number after the beginning of a scheduled lock time.

Feature interactions

Automatic Call Distribution

The Scheduled Electronic Lock feature does not support Automatic Call Distribution (ACD) sets, as CCOS does not support ACD sets.

Direct Inward System Access

Direct Inward System Access (DISA) numbers are not assigned to Scheduled Access Restrictions groups, so they are not affected by the SELK feature.

Electronic Lock Network Wide / Electronic Lock on Private Lines

The SELK feature supports Electronic Lock Network Wide / Electronic Lock on Private Lines. However, a scheduled lock is not supported over a network. Scheduled Electronic Lock must be configured and administered locally. Like SELK, these features obtain their restrictions from Scheduled Access Restrictions.

Message Intercept

When SELK locks a telephone set, Message Intercept (MINT) provides a different dial tone or announcement while the telephone set is locked.

Multi Tenant Service

If Scheduled Access Restrictions are applied to a tenant, the telephones in that tenant group follow the Scheduled Access Restrictions (unless the telephone belongs to a different SAR group).

Scheduled Access Restrictions

The Scheduled Access Restrictions (SAR) Permanent Disable, Active Lock, and Lock Disable FFCs have precedence over SELK.

Feature packaging

Scheduled Electronic Lock requires the following packages:

- Controlled Class of Service (CCOS) package 81
- Flexible Feature Code (FFC) package 139
- Scheduled Access Restrictions (SAR) package 162
- Message Intercept (MINT) package 163, if the Message Intercept function is required

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 88 – Configure the Scheduled Access Restrictions data block.

2 LD 10 – Configure the Scheduled Electronic Lock feature on an analog (500/2500 type) telephone set.

3 LD 11 – Configure the Scheduled Electronic Lock feature on a digital telephone on Remote Office.

4 LD 57 – Define Flexible Feature Codes for Scheduled Electronic Lock.

LD 88 – Configure the Scheduled Access Restrictions data block.

Prompt	Response	Description
REQ	NEW CHG	Create data block. Change existing data block.
TYPE	SAR	Scheduled Access Restrictions.
CUST	xx	Customer number, as defined in LD 15

SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15). ***Note:*** This prompt does not appear to a user with an LAO password.
SGRP	0-999	Scheduled Access Restrictions group number.
OFFP	1-8	Off-hour period number. Off-hour periods can overlap; the period that starts first has priority until that off-hour period is finished. All the prompts shown up to the ICR prompt repeat until you enter <CR>.
	<CR>	Go to the ICR prompt.
- STAR hh mm	hh mm	Start time. The current start time (hours and minutes) is printed individually after the prompt. Respond with the new start time.
- STOP hh mm	hh mm	Stop time. The current stop time (hours and minutes) is printed individually after the prompt. Respond with the new stop time.
- DAYS	d ... d	Respond with a new set of days to be used. Maximum of seven entries in the range of 1-7. Day 1 = Sunday, Day 2 = Monday, etc.
- COS		Off-hour period Class of Service.
	(UNR)	Unrestricted
	CTD	Conditionally Toll-Denied
	CUN	Conditionally Unrestricted
	FR1	Fully Restricted Class1
	FR2	Fully Restricted Class 2
	FRE	Fully Restricted
	SRE	Semi-restricted
	TLD	Toll Denied
- TGAR	0-(1)-15	Trunk Group Access Restriction.
- NCOS	0-99	Network Class of Service.
ICR	(NO) YES	Incoming Calls are Restricted.
LOCK	(1)-8	Indicates off-hour period to be used as the LOCK period. Default is Period 1.

LD 10 – Configure the Scheduled Electronic Lock feature on an analog (500/2500 type) telephone set.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	500	Analog (500/2500 type) telephone.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	x...x	ODAS Station Designator.
CUST	xx	Customer number, as defined in LD 15
...		
SCPW	xxxx	Station Control Password. SCPL must be configured in LD 15.
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.
...		
CLS	CCSA	Controlled Class of Service Allowed. CCSD = Controlled Class of Service Denied (default).
	SLKA	Scheduled Electronic Lock Allowed. SLKD = Scheduled Electronic Lock Denied (default).

LD 11 – Configure the Scheduled Electronic Lock feature on a digital telephone on Remote Office.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	aaaa	Digital telephone on Remote Office. aaaa = 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3901, 3902, 3903, 3904, 3905.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	x...x	ODAS Station Designator.
CUST	xx	Customer number, as defined in LD 15
...		
SCPW	xxxx	Station Control Password. SCPL must be configured in LD 15.
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.
...		
CLS	CCSA	Controlled Class of Service Allowed. CCSD = Controlled Class of Service Denied (default).
	SLKA	Scheduled Electronic Lock Allowed. SLKD = Scheduled Electronic Lock Denied (default).

LD 57 – Define Flexible Feature Codes for Scheduled Electronic Lock.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Provide FFC confirmation tone.
CODE	ELKA	New/change Electronic Lock Activate FFC.
ELKA	xxxx	Enter the new or changed Electronic Lock Activate FFC.

CODE	ELKD	New/change Electronic Lock Deactivate FFC.
ELKD	xxxx	Enter the new or changed Electronic Lock Deactivate FFC. ELKD must be different than ELKA.

Feature operation

During the period that a telephone set is locked, the user must enter the ELKD FFC to unlock the telephone set. The telephone set remains unlocked until either the user dials the ELKA FFC to manually lock the telephone set or the next scheduled lock occurs.

Secrecy Enhancement

Contents

This section contains information on the following topics:

Feature description 649
Operating parameters 650
Feature interactions 650
Feature packaging 651
Feature implementation 651
Feature operation 651

Feature description

This feature allows a warning tone to be applied to a three-way connection involving the source, destination and attendant if Warning Tone Allowed (WTA) Class of Service is available on both the source and destination sides. If the warning tone is denied on either the source or destination, these parties are automatically split. This applies to all calls handled by the attendant instead of only incoming network calls and attendant recalls with the original secrecy feature. The warning tone is always applied to a three-way connection.

There will be no connection established through the console with more than two parties, excluding the attendant, unless all parties have WTA Class of Service.

This feature also prevents any intelligible crosstalk on an attendant-held call or if the source (SRC) or destination (DEST) party is excluded.

Operating parameters

A connection is not established through the console if one of the parties, excluding the attendant, has warning tone denied Class of Service.

Feature interactions

AC15 Recall: Timed Reminder Recall

When the attendant answers an AC15 recall, the destination party is excluded from the connection. The attendant is connected to the source party and the excluded destination lamp is lit to show the exclusion of the destination party.

Attendant Break-In with Secrecy

The source and destination parties cannot be joined together on the attendants conference bridge if BKIS is active. This is consistent with the existing Break-In feature.

Attendant Recall

When the attendant answers a recall, the attendant is automatically connected to the destination party and the source party is excluded.

Semi-Automatic Camp-On

Secrecy and Enhanced Secrecy apply to Semi-automatic Camp-On recalls, with splitting taking place when the attendant answers the recall.

Secrecy

All functionalities of the Secrecy feature apply to the Secrecy Enhancement feature.

Slow Answer Recall Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Source Included when Attendant Dials

Source Included when Attendant Dials takes precedence over Secrecy and Enhanced Secrecy.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 15 – Select secrecy enhancement for customer:

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	FTR	Features and options Data Block
...		
OPT	(SYD) SYA EHS	Secrecy allowed Enhanced Secrecy allowed Secrecy denied

Feature operation

No specific operating procedures are required to use this feature.

Secretarial Filtering

Contents

This section contains information on the following topics:

Feature description 653
Operating parameters 653
Feature interactions 654
Feature packaging 655
Feature implementation 655
Feature operation 655

Feature description

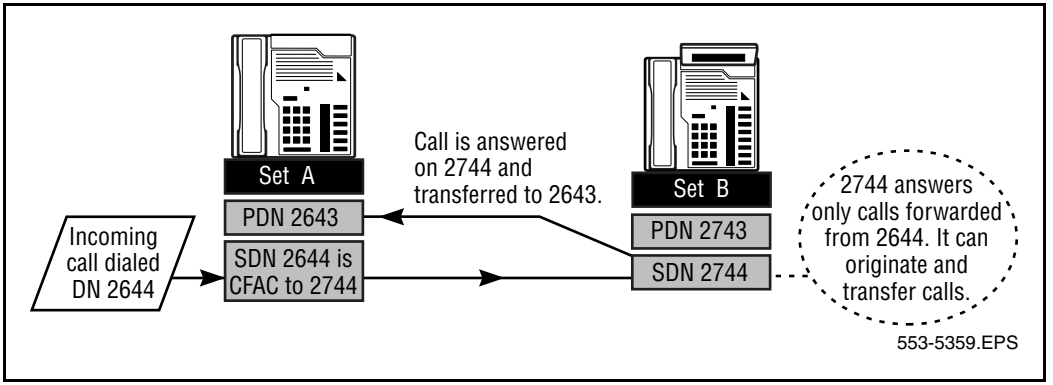
Secretarial Filtering is an application of Call Forward All Calls. It allows you to forward all calls to a second telephone. The user at the second telephone answers the forwarded calls and can choose to transfer the call back to you.

In the following example, a manager having a secondary Directory Number (DN) of 2644 forwards all calls arriving at that DN to a secretary's secondary DN 2744. Any call placed to DN 2644 is forwarded to the secretary at DN 2744. The secretary answers the call, decides that the manager should take the call, and transfers it back to DN 2643 (the prime DN). In this example, the manager receives only the calls originated or transferred by the secretary.

Operating parameters

Only the Directory Number (DN) designated as the Call Forward number can originate or transfer calls to the originally dialed DN.

Figure 11
Secretarial Filtering example



All Single Appearance DNs on the forwarded telephone are forwarded to the target DN.

A Multiple Appearance DN on the forwarded telephone is forwarded only if it is a Prime DN. A Multiple Appearance DN that is not the Prime DN rings at all appearances, including the forwarded telephone.

Feature interactions

Call Forward/Hunt Override Via Flexible Feature Code

The Secretarial Filtering feature is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Network Intercom

In a Secretarial filtering scenario, the secretary's BFS lamp also will reflect that the boss' set is busy if the boss is on a Hot Type I call.

Phantom Terminal Numbers (TNs)

If a Phantom TN is call forwarded to an existing telephone, and that telephone is used to call a DN on the Phantom TN, the call receives DCFW treatment.

Feature packaging

Secretarial Filtering is included in base system software. It is provided with Call Forward All Calls.

Feature implementation

This feature is enabled when Call Forward All Calls is enabled.

Feature operation

See the feature operation in the Call Forward All Calls module in this document.

Seizure Acknowledgment

Contents

This section contains information on the following topics:

Feature description 657
Operating parameters 658
Feature interactions 658
Feature packaging 658
Feature implementation 658
Feature operation 658

Feature description

Outgoing Ear and Mouth (E&M) Direct Inward Dialing (DID) or Direct Outward Dialing (DOD) trunks with an immediate start arrangement may require a seizure acknowledgment signal be received after a trunk seizure. This signal is an off-hook message. If the signal is not received within one second of the seizure, the trunk is software busied for three seconds, then dropped. The outgoing call then attempts to seize the next trunk in the sequence to complete the call. If the signal is received, the call is processed normally.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Seizure Acknowledgment feature described above.

The Seizure Acknowledgment feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Set Seizure Acknowledgement for a trunk route.

Prompt	Response	Description
REQ	aaaa	Request (aaaa = CHG, END, LCHG, NEW, OR OUT)
TYPE	RDB	Type of data block = RDB (Route data block)
CUST	xx	Customer number, as defined in LD 15
...		
ACKW	(NO) YES	Seizure acknowledgment signal (is not) is expected after seizure of this DID/DOD trunk.

Feature operation

No specific operating procedures are required to use this feature.

Selectable Conferee Display and Disconnect

Contents

This section contains information on the following topics:

Feature description 659
Operating parameters 668
Feature interactions 671
Feature packaging 678
Feature implementation 678
Feature operation 681

Feature description

The Selectable Conferee Display and Disconnect (SCDD) feature expands existing Conference Display functionality and provides Meridian Modular (Aries) set users with the capability to selectively drop any party that has been added to a conference. This feature provides Meridian Modular sets involved in a conference with the following two enhancements:

- Conference Count Display
- Selectable Conferee Disconnect

Note: The Selectable Conferee Display and Disconnect feature applies to Meridian Modular sets equipped with a display screen. The Meridian Modular set must be participating in a conference involving a total of at least three conferees.

Conference Count Display

Previously, only the elapsed time was shown on the display screen of a Meridian Modular set during a conference call. With Conference Count Display, however, the display screen of a Meridian Modular set also shows a count of the number of parties currently active in a conference call. This count includes every conferee involved in the active conference, whether a Meridian Modular set or not. The Conference Count Display is updated whenever a conferee is added to or disconnected from the active conference.

The Conference Count Display is activated at a set level by setting Class of Service to Conferee Display Count Allowed (CDCA). If Class of Service is set to Conferee Display Count Denied (CDCD), the display screen of the Meridian Modular set shows only the elapsed time, as per existing functionality.

The Conference Count Display is composed of three fields which are configured in the Customer Data Block. At least one of these fields must be configured for the Conference Count Display functionality to be in effect. The fields are as follows:

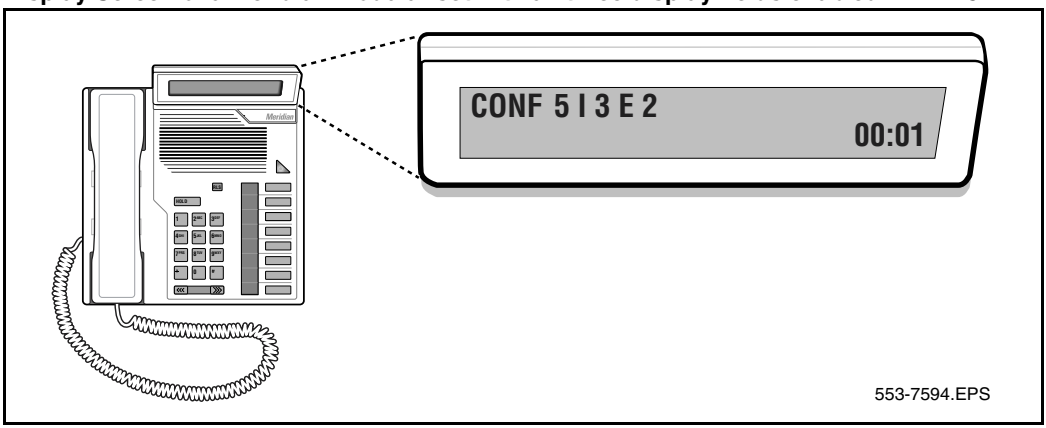
- The Total Conferees Count display field (CNFFIELD) shows the total number of parties involved in a conference (total internal conferees + total external conferees). The default mnemonic for this field on the display screen is “CONF”.
- The Total Internal Conferees Count display field (INTFIELD) shows the total number of conferees that are internal to the system. This includes analog (500/2500 type) sets, Meridian 1 proprietary sets, Attendant Consoles, and service trunks (such as Paging, Music, and Recorded Announcement) within the system. The default mnemonic for this field on the display screen is “I”.
- The Total External Conferees Count display field (EXTFIELD) shows the total number of conferees that are external to the system. This includes trunks that are connected to the system that can be configured on internal or external routes. The default mnemonic for this field on the display screen is “E”.

The mnemonics for each of the above fields can be modified to accommodate different languages or to save output time. This modification is performed by defining the CNF_NAME, INT_NAME, and EXT_NAME prompts in the Customer Data Block. The mnemonic for each of the three fields can be one to four characters in length.

When modifying the mnemonics for the three display fields, it is recommended that the real time impact be taken into consideration. Since each character, including spaces, is sent to a Meridian Modular set individually, a configuration with the maximum number of characters in each of the field headings (four characters each) affects the refresh time for each of the sets involved in the conference. This is especially important for conferences involving a large number of parties.

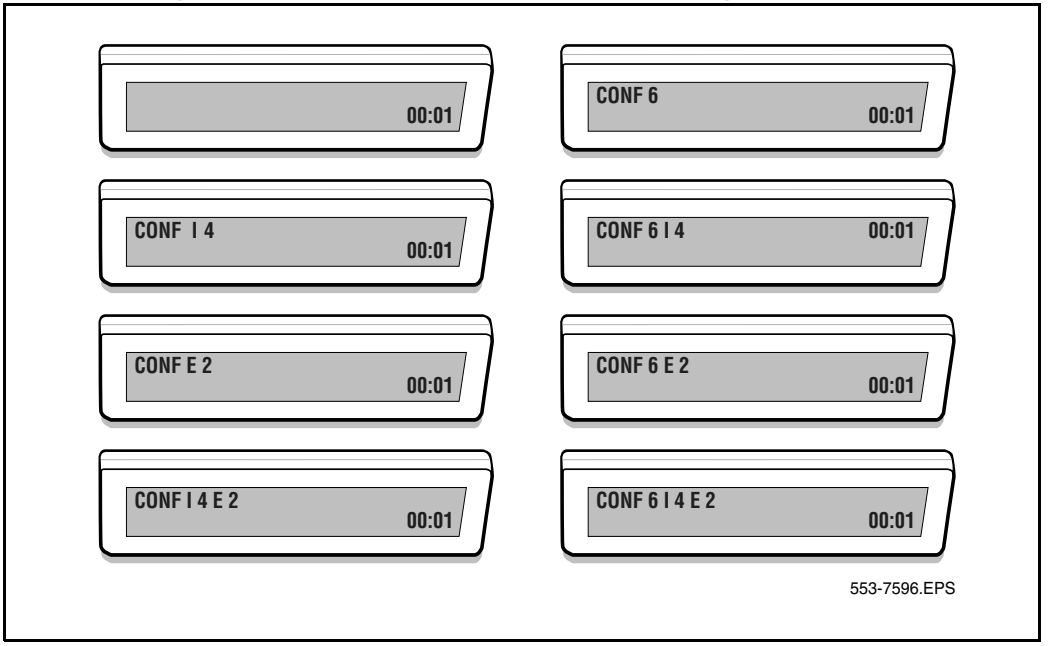
In Figure 12, a Meridian Modular set is involved in a conference consisting of five parties - three internal conferees and two external conferees. The Meridian Modular set has all three Conference Count Display fields (CNFFIELD, INTFIELD, and EXTFIELD) enabled in the Customer Data Block. Also, the Class of Service at a set level is set to Conferee Display Count Allowed (CDCA).

Figure 12
Display Screen of a Meridian Modular set with all three display fields enabled in LD 15



Eight possible Conference Count Display formats can be configured in the Customer Data Block, using a combination of the three Conference Count Display fields. Figure 13 shows the eight possible display formats for Conference Count Display. In this example, a conference has been established with a total of six conferees - four internal parties and two external parties.

Figure 13
Possible display formats of the active Conference Count Display

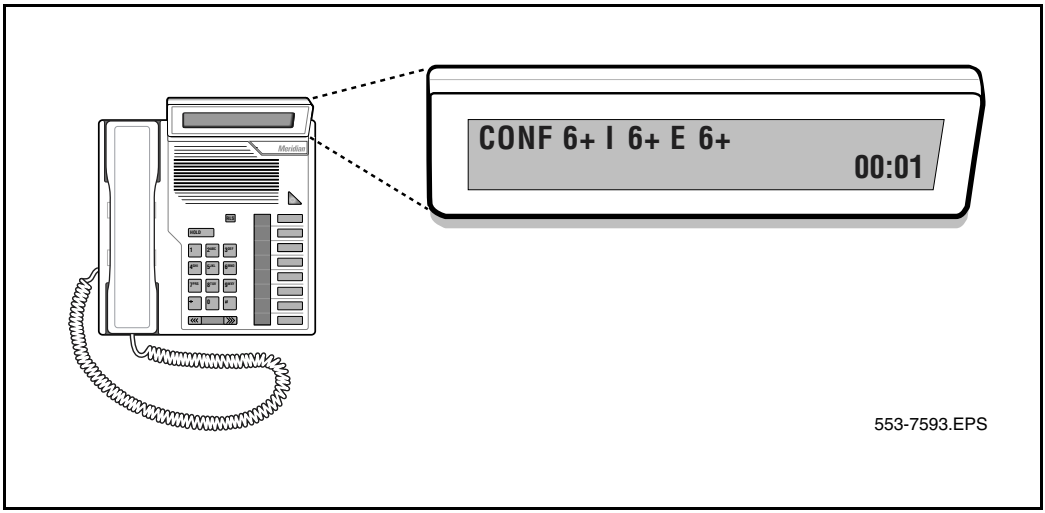


In Figure 13, the Total Conferees Count display field (CNFFIELD) is disabled in the left hand column. CNFFIELD is enabled in the right hand column. The formats in Row 1 have both the Total Internal Conferees Count display field (INTFIELD) and the Total External Conferees Count display field (EXTFIELD) disabled. In Row 2, the INTFIELD is enabled and the EXTFIELD is disabled. In Row 3, the INTFIELD is disabled and the EXTFIELD is enabled. INTFIELD and EXTFIELD are both enabled in Row 4.

Note: The Total Conferees Count display field name (CNF_NAME) is displayed when any of the CNFFIELD, INTFIELD, or EXTFIELD prompts are set to YES in the Customer Data Block.

Each display field on the screen of a Meridian Modular set shows a maximum conferee count of six. If the total number of conferees exceeds six, the Conference Count Display fields show "6+". In Figure 14, a Meridian Modular set is involved in a conference consisting of more than six external parties and more than six internal parties. Therefore, the Total Conferees Count also exceeds six.

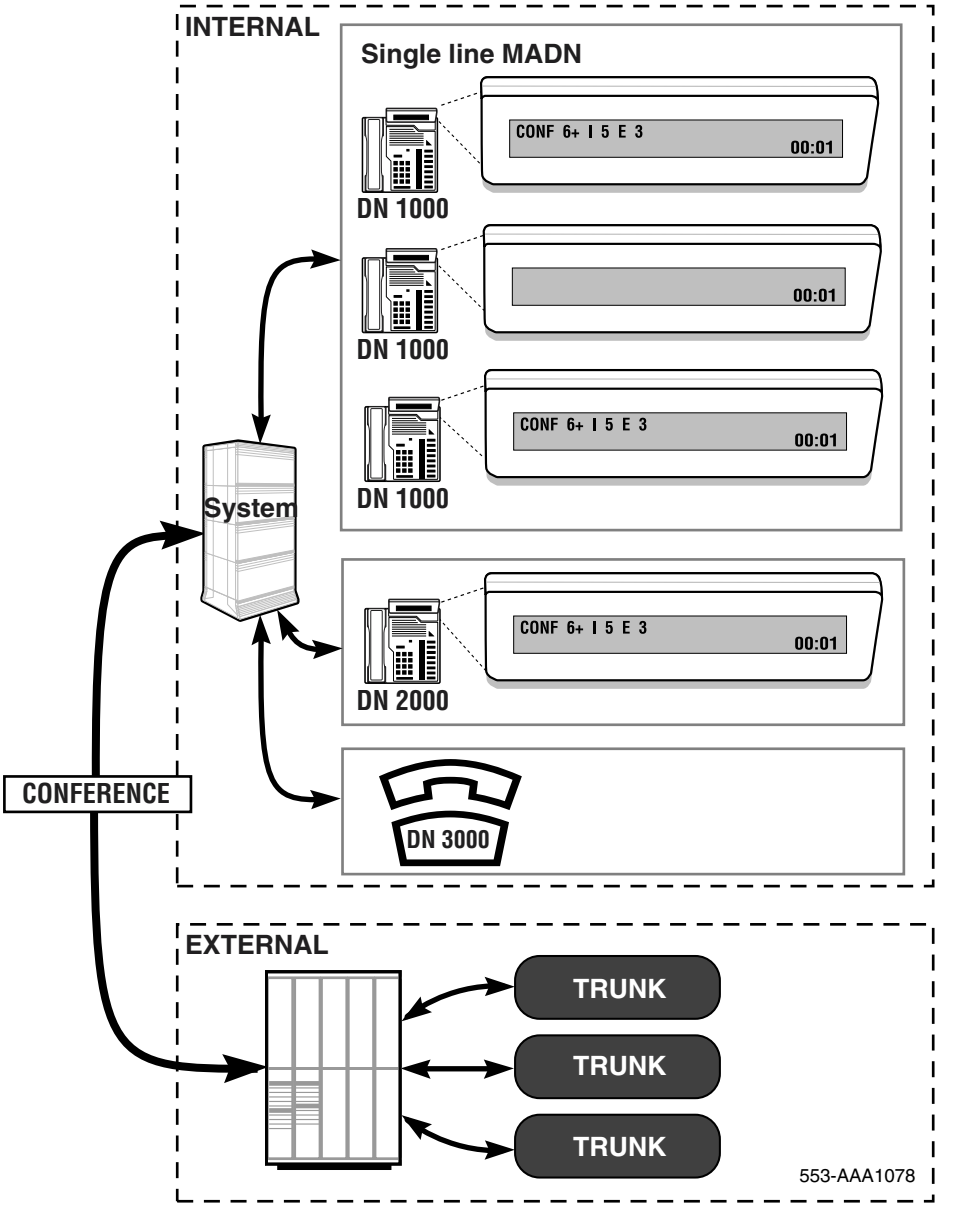
Figure 14
Display Screen of a Meridian Modular set involved in a conference where the total number of internal and external conferees exceeds six



In Figure 15 on page 665, five internal sets and three external trunks are involved in an active conference. The display screens of the Meridian Modular sets contain Conference Count Display information.

Referring to Figure 15, DN 1000 is a single line Multiple Appearance Directory Number (MADN) on three Meridian Modular sets. All three sets on DN 1000 are involved in the active conference (two of the sets entered the conference via Privacy Override). One of the sets on DN 1000 has Class of Service set to Conferee Display Count Denied (CDCD) in LD 11; therefore, its display screen shows only the elapsed time. All other Meridian Modular sets involved in the conference have Class of Service set to Conferee Display Count Allowed (CDCA). DN 2000, a Meridian Modular set, and DN 3000, an analog (500/2500 type) set, are also involved in the active conference. All three display fields are enabled in LD 15.

Figure 15
Example of a Conference Scenario involving both internal and external parties



Selectable Conferee Disconnect

With Selectable Conferee Disconnect, a Meridian Modular set user scrolls through a list of active conferees, using a Conferee Selectable Display (CSD) key. The CSD key is configured at a set level.

Selectable Conferee Disconnect is activated when the CSD key is pressed during an active conference. Every conferee involved in the conference, with the exception of the CSD key user, can be displayed one at a time on the CSD key user's screen.

When the CSD key is in use, the display format of each conferee follows the existing simple two party call display. The display shows the name and extension number of the conferee. If the conferee is on a trunk, the display shows the trunk group access code and the trunk member number.

Once the CSD key is activated, the key user can selectively disconnect a displayed conferee from the conference by pressing the active call key. The active call key is the key on which the conference is established. Also once the CSD key is activated, the CSD key user can cancel the Selectable Conferee Disconnect operation by pressing the Release key.

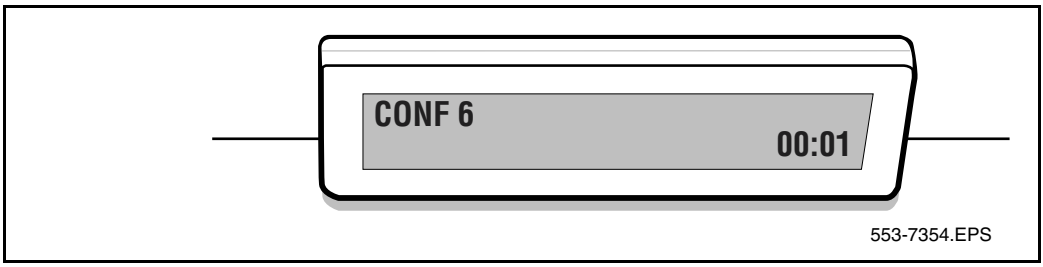
When the CSD key is pressed, the last conferee to join the active conference is displayed first. Subsequent pressing of the CSD key displays the other conferees in no particular order. With each press of the CSD key, the conferee list scrolls in a forward direction. Therefore, if the CSD key user scrolls past the desired party to be disconnected, repeated pressing of the CSD key brings the user to the desired party again.

The CSD key lamp is lit when the CSD key is activated. If, however, the displayed conferee cannot be disconnected, as in the case of an Attendant Console, the Key lamp flashes. The display screen remains unchanged and continues to show the same conferee on the display.

Figures 16 and 17 on page 667 and Figure 18 on page 668 show the display screen of Set A, a Meridian Modular set, as it displays and disconnects a selected conferee. Class of Service is set to CDCA in LD 11 and a CSD key is also configured in LD 11. Only the CNFFIELD is enabled in LD 15; therefore, only the Total Conferees Count is displayed.

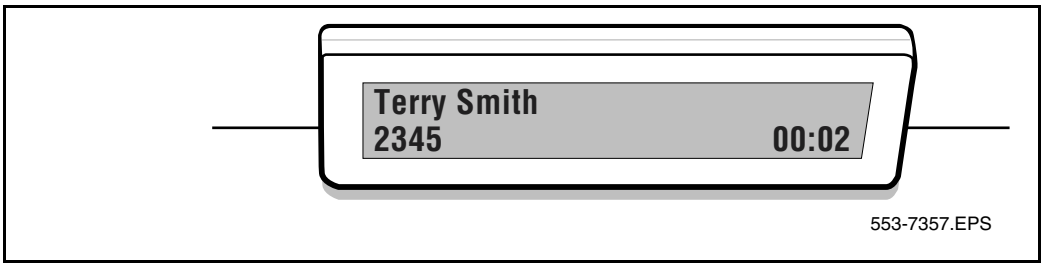
The display screen in Figure 16 shows that there is a total of six conferees involved in an active conference.

Figure 16
Display screen of Set A prior to the CSD key being activated



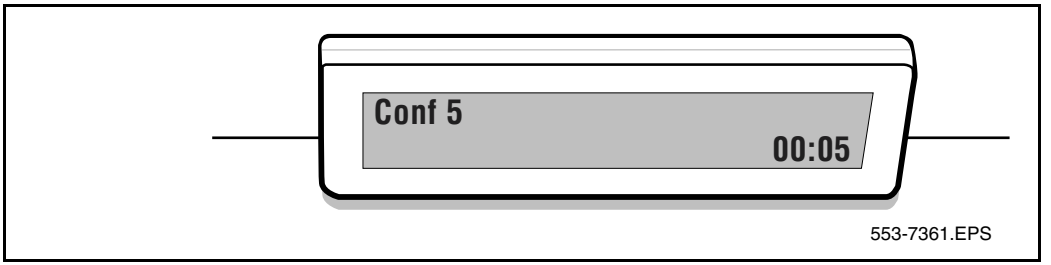
In Figure 17, Set A presses the CSD key and scrolls to conferee, Terry Smith.

Figure 17
Display screen of Set A when the CSD key is activated and a conferee to disconnect is selected



In Figure 18, Set A presses the active call key to disconnect Terry Smith. Set A's display screen is updated to show the new total conferee count status after the conferee is disconnected from the active conference. The Conference Count Displays of the other Meridian Modular sets involved in the conference are also updated. After the disconnection of Terry Smith, a total of five conferees remain active in the established conference.

Figure 18
Display screen of Set A after disconnecting the selected conferee



Operating parameters

The Selectable Conferee Display and Disconnect feature only applies to Meridian Modular sets that are equipped with a display screen. The Meridian Modular set must be participating in an active conference involving a total of at least three conferees.

Meridian Modular sets include M2008, M2016, M2616, M2216ACD1, and M2216ACD2 sets.

When conferees disconnect from an active conference, leaving only two parties in the conference call, the conference is usually converted to a simple two-party call. There are some situations, however, where the two remaining parties are still connected as a conference call. For instance, if either party is an Attendant Console or if both conferees are mixed sets with the same DN, the conference status is maintained.

The Selectable Conferee Display and Disconnect feature is not applicable to two party conferences.

Simple call display for the last two remaining parties in a conference is as per existing operation.

The method that is used to add a conferee to a conference does not affect Selectable Conferee Display and Disconnect. Some of these methods are: 3-party and 6-party Conference, Override, Attendant Barge-In, Attendant Break-In, and Bridging.

The two Selectable Conferee Display and Disconnect sub-features, Conference Count Display and Selectable Conferee Disconnect, have independent functionalities and operations.

Conference Count Display

Conferee Count Display is activated in LD 11 by setting Class of Service to Conferee Display Count Allowed (CDCA). At least one of the three Conference Count Display field options must also be enabled in the Customer Data Block.

In order for Class of Service to be set to Conferee Display Count Allowed (CDCA) in LD 11, the Automatic Digit Display (ADD) or the Delay Display (DDS) Class of Service must first be set.

A display screen with only the elapsed time showing (existing functionality) can be configured if all three Conference Count Display field options are set to NO in LD 15 or if Class of Service is set to Conferee Display Count Denied (CDCD) in LD 11.

Selectable Conferee Disconnect

Selectable Conferee Disconnect is activated by defining a Conferee Selectable Display (CSD) key in LD 11. Prior to defining the CSD key, however, Automatic Digit Display (ADD) or Delay Display (DDS) Class of Service must be set in LD 11.

Only one CSD key can be configured per Meridian Modular set.

The CSD key can only be used during an active conference call.

Each conferee (internal and external) is displayed to the CSD key user following the existing simple two-party call display. No changes are made to the features that supply and/or display the conferee's data. Some of these features are: Call Party Name Display, Calling Party Privacy, Dialed Number Identification Service, Digit Display, Display of Calling Party Denied, and ISDN Calling Line Identification.

For the display of a conferee that is on a trunk, the specific terminating set may not be shown. Therefore, when several trunks are involved in a conference, it recommended that a record be kept of what party joins the conference on what trunk.

When a conferee uses the CSD key, the displays (if any) on the other conferee sets are not changed. Only the CSD key user can see the list of conferees.

After the CSD key is pressed, only the active call key, Release (RLS) key, or CSD key can be used. All other input is ignored.

When the CSD key is activated, if the CSD key user goes on-hook, the key user is disconnected from the call instead of the displayed conferee.

This feature does not support the use of a confirmation tone as indication that a conferee has been disconnected from the active conference.

If the system initializes or sysloads during an active conference, the conference is torn down as per existing functionality. If the CSD key is active when the system initializes or sysloads, then the key operation is canceled.

When the last party to join the conference uses the CSD key, the active conferee list has no particular order. This is because the last conferee to join the conference is the only conferee to be displayed with any priority. In this case, the last party to join the conference is the CSD key user, and the CSD key user is never displayed.

If the last conferee to join the conference is disconnected, then the next scan of the active conferee list has no particular order. The order of inclusion of each conferee is not maintained or stored beyond the last conferee to join the conference.

A conferee can be disconnected from the active conference via the CSD key at any time during the conference call.

When a key or feature key is pressed, the active conference display is replaced. The Conference Count Display is not restored until a conferee is added to or disconnected from the conference, thereby updating the conferee count. If the conference is placed on hold and then restored, the Conference Count Display appears once again.

Feature interactions

Attendant

When the CSD key is activated, the Attendant Console can be displayed as a conferee in the active conference. The CSD key cannot be used to disconnect an Attendant Console from the conference. Only the Attendant Console can release itself from a conference call.

An attempt to disconnect the Attendant Console via the CSD key causes the CSD key lamp to flash. To recover from the flashing CSD key lamp, the key user presses the Release key to cancel the CSD key operation or presses the CSD key again to scroll to the next conferee.

Attendant Barge-In

When an attendant barges into a conference, the conferees are separated. The conferees connected through the trunk that is being verified are placed on the destination (DEST) side and do not include the attendant. The other conferees are conferenced on the Source (SRC) side and include the attendant. However, all parties can communicate with each other.

Once a conference is established on the SRC and/or DEST side, the CSD key is operable. The CSD key, however, cannot be used to disconnect an attendant.

Attendant Break-In

An attendant receives an urgent call and dials the destination DN which is busy. The attendant places the urgent call on hold and then breaks into the active call by using the Break-In key. The destination DN disconnects from the current active call so that the attendant can extend the urgent call.

If the attendant breaks into a simple call, a three-party conference is established including the attendant. Once the conference is established, the new Conference Count Display is not shown. Instead, the displays on the two sets show the attendant information.

If the attendant breaks into a conference call, the attendant is added to the existing conference. Once a conference is established, involving the attendant, the new Conference Count Display is not shown. Instead, the displays of the sets show the attendant information. When the Destination DN disconnects from the active conference, the urgent call is extended to the destination DN. If the remaining parties can form a conference, the Conference Count Display is shown on those sets.

Once a conference is established, the CSD key can be used. However, the urgent call is not shown as a conferee. The attendant is shown as a conferee, but the CSD key cannot be used to disconnect an attendant.

Attendant Administration

Attendant Administration (AA) is modified in order to print the Conferee Selectable Display key when found through LD 71. AA cannot be used to configure a CSD key.

Meridian Integrated Conference Bridge

The Selectable Conferee Display and Disconnect feature does not change the functionality of Meridian Integrated Conference Bridge.

Automatic Call Distribution

An Automatic Call Distribution (ACD) agent or supervisor can activate Conference and No Hold Conference. If the ACD set is a Meridian Modular set equipped with a display and a CSD key, then the Selectable Conferee Display and Disconnect feature can be used.

Agent Observe

Selectable Conferee Display and Disconnect does not change the functionality of the ACD Agent Observe feature. While in the observe mode, the ACD supervisor is not part of the conference. Thus, the active conference count does not include the ACD supervisor. The Conference Count Display is not shown on the ACD supervisor's set. When the CSD key is activated, the ACD supervisor is not shown in the active conferees list.

ACD Agent Features

It is recommended that the CSD key not be assigned to agents' sets, as the CSD key can be used to disconnect a supervisor.

Alternate Call Answer

When ACAA = YES in LD 23, an agent can put an active Individual DN (IDN) call on hold and then press the In-Calls key to return to the idle agent queue in order to take the next call. If the agent activates Call Join, a conference is established with the agent, the IDN call, and the ACD call.

With Alternate Call Answer, once there is an active conference established with the ACD agent, an IDN call, and the ACD call, the Selectable Conferee Display and Disconnect feature is applicable.

Agent and Supervisor Communication

When the ACD agent is active in a simple call with an ACD caller and wishes to include the ACD supervisor in the call, the ACD agent presses the Answer Supervisor (ASP) key. The supervisor answers by pressing the Agent (AGT) key. In order to finish this operation, the agent presses the ASP key once the supervisor has answered.

When an ACD agent is active in a conference call with an ACD caller, the supervisor cannot be added to the conference via the ASP key.

With Agent and Supervisor Communication, once there is an active conference established, the Selectable Conferee Display and Disconnect feature is applicable.

Emergency Key

The Emergency Key (EMR) feature enables the ACD agent to conference an ACD supervisor and, optionally, a recording device for customer-defined emergencies or sensitive situations.

When the EMR key is activated, the recording trunk is not considered a member of the conference. When the CSD key is activated, the recording trunk is not included in the active conferees list. The total number of conferees on the Conference Count Display does not include the recording trunk.

ACD Display Enhancement

With the ACD Display Enhancement, the Not Ready (NRD) key cannot be pressed when using the Conference key. When a conference is established and the NRD key is pressed, the conference call is disconnected. In this case, the NRD key lamp is lit, and the ‘NOT READY’ screen is displayed.

When the CSD key is active, the NRD key cannot be used. Pressing the NRD key is ineffective.

ACD In-Calls Key

When a conference is established on the ACD In-Calls key, the In-Calls key is used to drop a desired conferee when the CSD key is activated. The Position Identification (POS ID) of each ACD set involved in the conference is displayed when the CSD key user scrolls through the active conferees list.

Application Module Base

The Selectable Conferee Display and Disconnect feature uses the existing messaging to disconnect a conferee. The messaging to disconnect a conferee is the same as though the conferee has gone on-hook or has pressed the Release (RLS) key to disconnect themselves from the conference.

Automatic Hold

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Automatic Hold feature. Once a conference is established on the active DN key, the Selectable Conferee Display and Disconnect feature is applicable.

Basic Rate Interface

The Selectable Conferee Display and Disconnect feature is not supported on BRI sets. However, if a conferee involved in the active conference is on a BRI set, its information is shown when the CSD key is activated.

Bridging

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Bridging feature.

With the Bridging feature, the same DN can appear on up to eight single-line sets. Any appearance of the MADN can enter a call by going off-hook. When a conference with three parties is created through Bridging, there are only two active DNs in the conference call. As long as there are only two different DNs in the bridged conference call, the displays on the sets show the information of the other DN involved in the call, not the Conference Count Display information. In this case, however, the CSD key can be used, as more than three conferees are active in the conference call.

Once there are more than two different DNs in the conference call, the Conference Count Display shows the count of the conferees. Once a conference is established, the CSD key is applicable.

Conference

The Selectable Conferee Display and Disconnect feature does not change the functionality of Conference, except for the new active conference display. Conference calls can include calls on the following key types: Single Call Arrangement DN (SCN, SCR), Multiple Call Arrangement DN (MCN, MCR), ACD In-Calls (ACD DN), Private Line Ringing and Non-ringing (PLN, PLR), Hotline (HOT), Call Waiting (CWT), Voice Call (VCC) and Dial Intercom (DIG).

Conference Control

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Conference Control feature.

Digitone Receiver

The Selectable Conferee Display and Disconnect feature does not treat the Digitone Receiver (DTR) as a conferee when it appears on the conference loop since it appears only temporarily to provide the tone service.

Display Key

While in a conference call, the Display (DSP) key can be used to obtain information. However, the Display key is blocked when the CSD key is active.

DNIS Across Call Modifications

When a CSD key user scrolls through the list of conferees during a DNIS call, the DNIS information is displayed.

End-to-End Signaling

The Selectable Conferee Display and Disconnect feature does not block End-to-End Signaling (EES) or dialing digits while the CSD key is active.

Group Call

The Selectable Conferee Display and Disconnect feature is only applicable to the originator of a Group Call involving three or more active parties. The active conference display is not shown until a redisplay of the Group Call originator's screen is needed.

Hold

With the Selectable Conferee Display and Disconnect feature, when a Meridian Modular set equipped with display is involved in a conference, its display shows the Conference Count Display. If a Meridian Modular set puts the conference on hold by pressing the Hold key, the active DN key lamp flashes, and the display is cleared during the held operation. The Conference Count Display is restored upon completion of the held operation. The active DN key is pressed to restore the held conference call.

Meridian Link

The Selectable Conferee Display and Disconnect feature uses existing messages sent over the Meridian Link in order to provide the Conference Count Display and the Selectable Conferee Disconnect functionality.

No Hold Conference

The Selectable Conferee Display and Disconnect feature does not change the No Hold Conference (NHC) functionality. The Selectable Conferee Display and Disconnect feature is applicable to conferences created by No Hold Conference.

Override

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Override (OVR) feature. The Conference Count Display is not shown for an Override conference, as the Override display is shown instead. The CSD key, however, can be used to disconnect conferees in an Override conference.

Priority Override

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Priority Override (POVR) feature. The Conference Count Display is not shown for a POVR conference, as the Priority Override display is shown instead. The CSD key can, however, be used to disconnect conferees involved in a POVR conference.

Privacy

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Privacy feature. With Privacy enabled, only one appearance of a single line MADN can participate in a conference call. This appearance is included in the conferee counts.

Privacy Override

The Selectable Conferee Display and Disconnect feature does not change the operation of the Privacy Override (POA) feature.

A Meridian 1 proprietary set with Privacy Override Allowed (POA) Class of Service can bridge into an established call on a single line MADN. When a conference with three parties is created through Privacy Override, there are only two active DNs in the conference call. As long as there are only two different DNs in the POA bridged conference call, the displays on the sets show the information of the other DN involved in the call, not the Conference Count Display information. In this case, however, the CSD key can be used, as more than three conferees are active in the conference call.

Once there are more than two different DNs involved in the active conference call, the Conference Count Display shows the count of conferees. The conferees that are added to the conference through POA are included in the Conference Count Display totals. Once a conference is established, the CSD key is applicable.

Tone and Digit Switch

The Selectable Conferee Display and Disconnect feature does not treat the Tone and Digit Switch (TDS) as a conferee when it appears on the conference loop, as it appears only temporarily to provide tone service.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Configure the Conference Count Display Format for the customer.
2. LD 11 – Set the Conferee Display Count Allowed (CDCA) Class of Service for Meridian Modular sets.
3. LD 11 – Configure a Conferee Selectable Disconnect (CSD) key for Meridian Modular sets.

LD 15 – Configure the Conference Count Display Format for the customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
CONF_DSP	YES	Change Conference Count Display configurations. NO = Do not change Conference Count Display configurations (default). To prompt for further conference display options, CONF_DSP must be set to YES.
- CNFFIELD	(NO) YES	Total Conferees Count display field (disabled) enabled.

- CNF_NAME	(CONF) aaaa	Total Conferees Count display field name. Enter 1-4 alphanumeric characters to replace the existing name. The Total Conferees Count display field name is displayed when any of the CNFFIELD, INTFIELD, or EXTFIELD prompts are set to YES.
- INTFIELD	(NO) YES	Total Internal Conferees Count display field (disabled) enabled.
--INT_NAME	(I) aaaa	Total Internal Conferees Count display field name. Enter 1 to 4 alphanumeric characters to replace the existing name.
- EXTFIELD	(NO) YES	Total External Conferees Count display field (disabled) enabled.
--EXT_NAME	(E) aaaa	Total External Conferees Count display field name. Enter 1 to 4 alphanumeric characters to replace existing name.

LD 11 – Set the Conferee Display Count Allowed (CDCA) Class of Service for Meridian Modular sets.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type where xxxx is: 2008, 2016, 2216, or 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		

CLS	ADD DDS (CDCA)	Automatic Digit Display. Delay Display. With CLS = DDS, the display is activated after the call is answered. CLS must be set to either ADD or DDS prior to setting CLS = CDCA or CDCD. Conferee Display Count Allowed (default) CDCD = Conferee Display Count Denied. CDCD option sets a blank display screen during a conference call.

LD 11 – Configure a Conferee Selectable Disconnect (CSD) key for Meridian Modular sets.

Prompt	**Response**	**Description**
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	xxxx	Telephone type where xxxx is: 2008, 2016, 2216, and 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CLS	ADD DDS	Automatic Digit Display. Delay Display. With CLS = DDS, the display is activated after the call is answered. CLS must be set to either ADD or DDS prior to configuring a CSD key.
KEY	xx CSD	Conferee Selectable Display key. To remove the CSD key, set the KEY prompt to xx NUL, thereby disabling Selectable Conferee Disconnect.

Feature operation

Viewing the list of active conferees

To view the list of active conferees:

- Press the Conferee Selectable Display (CSD) key to view the list of active conferees. Continue to press the CSD key to view each conferee. The CSD key lamp is lit. The displays on the other Meridian Modular sets involved in the conference are not changed.
- Press the Release key to cancel the Selectable Conferee Disconnect operation. None of the conferees are disconnected. The CSD key lamp is dark. The Conference Count Display returns if it is enabled. The original conference call remains active throughout this operation.

Disconnecting one conferee

To disconnect a conferee using the CSD key:

- Press the CSD key repeatedly until the conferee that is to be disconnected is displayed on the screen. The CSD key lamp is lit. The displays on other Meridian Modular sets are not changed.
- Press the active call key (the key on which the active conference is established). The displayed conferee is disconnected. The CSD key lamp is dark. The Conference Count Display returns, if enabled, showing the revised total count of conferees. The original conference call remains active throughout this operation.

Disconnecting more than one conferee

In order to disconnect more than one conferee, follow the steps for disconnecting one conferee. Each conferee must be disconnected separately.

> ***Note:*** When two CSD key users wish to drop different conferees (but not each other), each CSD key user can initiate the Selectable Conferee Disconnect operation and disconnect the selected conferee. If enabled, the Conference Count Displays on the Meridian Modular sets are revised once each Selectable Conferee Disconnect operation has concluded successfully.

Disconnecting the same conferee

Two Meridian Modular sets (Set A and Set B), both equipped with a CSD key, wish to disconnect the same conferee. The Set that presses the active call key first is successful in disconnecting the conferee. If Set A is the first set to press the active call key, its Conference Count Display is updated with the revised total count of conferees. The Conference Count Display of all other Meridian Modular sets, with the exception of Set B, are also updated. Set B's Conference Count Display is updated when it presses the active call key or when it presses the Release key to end the operation.

Verifying that a conferee has been disconnected

To verify that a conferee has been disconnected:

- View the list of conferees using the CSD key, and note whether or not the disconnected conferee is still listed.
- Check that the CSD key lamp is dark. This indicates that the Conferee Selectable Disconnect operation is complete.
- Check that the total count of conferees on the Conference Count Display has been revised on the display screen.
- If the conferee is disconnected and only two parties remain, a simple call situation is established. Therefore, the displays are updated accordingly.

Canceling the Selectable Conferee Disconnect operation

To cancel Selectable Conferee Disconnect operation at any time, press the Release key when the Conferee Selectable Disconnect operation is in progress. When the Release key is pressed, none of the conferees are disconnected, the CSD key lamp is dark, and the Conference Count Display returns (if enabled). The original conference call remains active throughout this operation.

Disconnecting from an active conference

To disconnect yourself from an active conference, press the Release key or go on-hook. In this case the original conference call remains active, as long as a supervised conference situation remains.

Selectable Directory Number Size

Contents

This section contains information on the following topics:

Feature description 683
Operating parameters 683
Feature interactions 684
Feature packaging 684
Feature implementation 684
Feature operation 684

Feature description

The Selectable Directory Number Size feature allows a user to define the number of digits that must be received on a Direct Inward Dialing (DID) route before the end of dialing (EOD) is reached. If the required number of digits is not received when the EOD timer expires, a TRK137 message is sent to print and the trunk is locked out.

The DN size can be specified from one to seven digits, or as zero which will not consider the number of digits dialed in the sequence.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Seizure Acknowledgment feature described above.

The Seizure Acknowledgment feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

LD 16 – Set limits for the Selectable Directory Number Size feature.

Prompt	Response	Description
REQ	aaaa	Request (aaaa= CHG, END, LCHG, NEW, or OUT
TYPE	RDB	Type of data block = RDB (Route data block)
CUST	xx	Customer number, as defined in LD 15
...		
DNSZ	(0)-7	Number of digits expected on DID routed; 0 indicates no fixed number.

Feature operation

No specific operating procedures are required to use this feature.

Semi-Automatic Camp-On

Contents

This section contains information on the following topics:

Feature description 685
Operating parameters 686
Feature interactions 687
Feature packaging 688
Feature implementation 688
Feature operation 689

Feature description

This feature allows a Camp-On call to recall to the attendant instead of ringing the called party when the called party becomes available. The called party can originate calls but cannot receive any other calls. Other incoming calls to this DN will receive a busy indication. If the called party originates another call when the attendant attempts to present the Camp-On call, the attendant receives busy tone and can initiate Camp On again or release the call.

When an attendant extends a call to a desired party that is busy, the attendant can activate Semi-automatic Camp-On by pressing the Semi-automatic Camp-On (SACP) key. This causes the call to be camped-on to the desired party, and recalled to the attendant when the desired party becomes idle, rather than rung through to the desired party.

Recall to Same Attendant must be allowed, otherwise the recall is routed to the first available attendant. The attendant display shows the calling-party DN and the party to which the call is camped-on. If the attendant, or all attendants in a multiple-console environment, are busy then the recall is placed in the attendant queue.

Meanwhile, incoming calls to the desired party receive busy treatment. The desired party, however, is still able to make calls. After receiving the recall, the attendant can ring the desired party by pressing the SACP key. The attendant may release the call while it is ringing, or hold the call until it is answered. If the desired party has made another call while the attendant tries to present the recall, the attendant may Camp-On the recall to the desired party by pressing the SACP key.

Operating parameters

The same operating parameters apply as for Camp-On.

Semi-automatic Camp-On is mutually exclusive with the Call Waiting feature. Thus, Attendant Consoles configured with Semi-automatic Camp-On will not work if Call Waiting has been defined.

Semi-automatic Camp-On can be configured for individual or all Camp-On occurrences.

Semi-automatic Camp-On is not available with Network Attendant Service. If the attendant tries to apply Semi-automatic Camp-On to a station at a remote node, the SACP lamp flashes to indicate that Semi-automatic Camp-On is not allowed. The attendant has to press the SACP key again to deactivate the feature, and be allowed to activate it under normal operation.

Semi-automatic Camp-On is not supported during Night Service or Enhanced Night Service. Calls that were camped-on by Semi-automatic Camp-On during normal hours ring through to the desired party, when idle, and do not recall to the attendant.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature uses the SACP key to activate a blocking attempt, but the Attendant Blocking of DN feature is only valid on the source side of the Attendant Console. The Semi-automatic Camp-on feature is only valid on the destination side of the Attendant Console.

To have the Attendant Blocking of DN feature available and not the Semi-automatic Camp-on feature, a new response to the SACP prompt has been introduced in LD 15. Prompt SACP = NO means the Semi-automatic Camp-on feature is not available even if the SACP package is equipped and an SACP key exists on the Attendant Console. To have the Semi-automatic Camp-on feature available the SACP prompt must be answered with SNGL or ALL which have the same meanings as before.

Attendant Break-In

The attendant can Break-In to an established call and apply Semi-automatic Camp-On to the desired party. The attendant may press the SACP key before or after the Break-In.

Call Forward/Hunt Override Via Flexible Feature Code

Semi-Automatic Camp-On can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When encountering a busy set, it is possible to activate SACP, if it is applicable.

Incoming calls during recall

During Semi-automatic Camp-On, when the desired party becomes idle and the camp-on is recalled to the attendant, the desired party appears busy to incoming calls. The DN of the desired party is displayed as busy on the Busy Lamp/Enhanced Busy Lamp display.

Periodic Camp-On Tone

Periodic Camp-On Tone stops when the camped-on call is recalled to the attendant.

Secrecy and Enhanced Secrecy

Secrecy and Enhanced Secrecy apply to Semi-automatic Camp-On recalls, with splitting taking place when the attendant answers the recall.

Source Included when Attendant Dials

The source remains included while the attendant dials the destination.

Feature packaging

This feature requires Semi-automatic Camp-On (SACP) package 181.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure Semi-automatic Camp-On for the customer.

2 LD 12 – Configure an SACP key on the Attendant Console.

LD 15 – Configure Semi-automatic Camp-On for the customer.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	ATT	Attendant Data Block.
...		
RTSA	RSAA	Recall To Same Attendant Allowed.
SACP	 (NO) SNGL ALL	Semi-automatic Camp-On. Semi-automatic Camp-On not allowed. Enable Semi-automatic Camp-On on a per-call basis. Enable Semi-automatic Camp-On for all occurrences. SACP keys must be defined on all Attendant Consoles which are to make use of the feature.

LD 12 – Configure an SACP key on the Attendant Console.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ATT	Attendant data block
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
KEY	xx SACP	Key Number, Semi-automatic Camp-On

Feature operation

When an attendant extends a call to a desired party who is busy, the attendant can activate Semi-automatic Camp-On as follows:

- Press the SACP on the Attendant Console.
  The call is camped on the desired party.
- The display on the Attendant Console shows the calling party's DN, and the party to which the call is camped on (the desired party).
- The desired party becomes idle.
  The call is recalled to the attendant.
- To ring the desired party after receiving the recall, press the SACP on the Attendant Console again.

Recall timing on Camp-On calls

When any station extends an external call, recall timing will be initiated if the call is camped on to a busy station.

The recall timing will start from the moment that the extending station "releases" the call. The value of the recall timer is set by the prompt RTIM in the Customer Data Block (LD 15).

At the recall, the camped on call will be routed to the attendant. If the attendant is in Night Service, Night treatment is given; if NAS routing is active, the call will be routed according to the NAS configuration.

Standalone case

When the recall to the attendant occurs, the Camp-On is canceled. If the attendant is busy during the recall, the recall will be queued.

Network case

When the recall occurs and the attendant has answered the recall, the call will still be camped on to the desired party. If during the recall the attendant is busy, the recall will be queued.

Series Call

Contents

This section contains information on the following topics:

Feature description 691
Operating parameters 692
Feature interactions 692
Feature packaging 693
Feature implementation 693
Feature operation 694

Feature description

The Series Call feature causes a source call (either an attendant-answered incoming call, or an attendant-originated trunk call), that has been extended to an internal destination party, to be recalled to the attendant when the destination party hangs up. The attendant can then extend the source call to another destination party. This feature enables a caller to talk to more than one party without having to disconnect and call again (Recall to Same Attendant must be allowed, otherwise the recall is routed to the first available attendant). This process can be repeated for as many destinations as requested by the caller.

A Series Call is canceled if one of the following occurs:

- the attendant presses the Series Call (SECL) key while the associated lamp is lit
- the attendant extends the source to a trunk while the SECL lamp is lit
- the attendant enters Night Service after extending the call and prior to receiving the recall
- the destination is call forwarded to a trunk, or
- the source disconnects.

Operating parameters

This feature only applies when the destination party is internal. If the attendant dials a DN that is not internal, the SECL key will flash to indicate that the feature cannot be invoked.

The source can only be extended to an internal party.

Feature interactions

Attendant Position Busy

If the attendant activates Position Busy while a Series Call is active, the recall occurs to the next available attendant.

Call Detail Recording

With Call Detail Recording, a start record is generated when a source Periodic Pulse Metering call is answered and marked as a Series Call by the attendant, and an end record is generated when the attendant releases the call. No intermediate records are generated.

Night Service

If the attendant extends a Series Call and goes into Night Service before it recalls to the attendant, the call recalls to the night DN and Series Call treatment is canceled.

Timed Reminder Recall

With Timed Reminder Recall, if the attendant extends a Series Call during Camp-on, Call Waiting, or ringing, the SECL lamp goes dark.

Feature packaging

This feature requires Series Call (SECL) package 191.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 12 – Configure Series Call for each Attendant Console.
2. LD 15 – Configure Recall to Same Attendant for the customer.

LD 12 – Configure Series Call for each Attendant Console.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ATT	Attendant data block
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
KEY	xx SECP	Key Number, Series Call

LD 15 – Configure Recall to Same Attendant for the customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Change Attendant Console options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- RTSA	(RSAD) RSAA	Recall (Denied) Allowed to Same Attendant.

Feature operation

The attendant designates the source call as a Series Call by pressing the Series Call (**SECL**) key. The **SECL** key may be pressed by the attendant while dialing, talking to the destination party, or while a call is ringing. The associated key lamp remains lit until the Series Call is canceled. If the attendant tries to extend a call to an external station, the SECL lamp flashes. The attendant has to press the **SECL** key to cancel the Series Call, and extend the call as a standard call extension.

Set-Based Administration Enhancements

Previously, Set-Based Administration was a feature available on Small Systems that simplified system installation and administration by enabling a set to be used to perform several administrative and maintenance procedures. With the Set-Based Administration Enhancements feature, Set-Based Administration is now available for all system types. In addition, enhancements are provided to the existing capabilities on Small Systems.

For more information about the Set-based Administration Enhancements feature, please see *Set-Based Administration* (553-3001-303).

Short Buzz for Digital Telephones

Contents

This section contains information on the following topics:

Feature description 697
Operating parameters 697
Feature interactions 698
Feature packaging 698
Feature implementation 698
Feature operation 698

Feature description

When a call is presented to a digital telephone that is off-hook, a buzz tone is given. The duration of this secondary buzz is shortened from two seconds to an average of 0.8 seconds, with a minimum length of 0.5 seconds and a maximum length of one second.

Operating parameters

Short Buzz for digital sets does not change the buzz tone given to Automatic Call Distribution (ACD) telephones on the In-calls key.

Feature interactions

Directory Number Delayed Ringing

If a set is defined with Directory Number Delayed Ringing (DNDR) delay and there is an incoming call to another SCN/MCN DN key on the same set, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Group Call

The special three-second buzz for Group Call is not affected by this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Single Appearance Directory Number

Contents

This section contains information on the following topics:

Feature description 699
Operating parameters 699
Feature interactions 699
Feature packaging 700
Feature implementation 700
Feature operation 701

Feature description

A Single Appearance Directory Number (SADN) can be assigned to any type of telephone.

Operating parameters

A Single Appearance Directory Number (SADN) is a DN that appears only once within a customer group.

Feature interactions

Directory Number Expansion

The DN can have up to seven digits if the Directory Number Expansion package is equipped.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Assign a Directory Number.

2 LD 11 – Assign Single Appearance Directory Number keys.

LD 10 – Assign a Directory Number.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Analog (500/2500 type) telephone.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
DN	x...x	Directory Number. Up to four digits; up to seven digits with DNXP package 150.

LD 11 – Assign Single Appearance Directory Number keys.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx SCN yyy...y	Add a single-call non-ringing DN key, where: xx = key number, and yyy...y = DN.
	xx SCR yyy...y	Add a single-call ringing DN key, where: xx = key number, and yyy...y = DN.

Feature operation

No specific operating procedures are required to use this feature.

Single-digit Access to Hotel Services

Contents

This section contains information on the following topics:

Feature description 703
Operating parameters 703
Feature interactions 704
Feature packaging 704
Feature implementation 704
Feature operation 706

Feature description

In hospitality applications, it is desirable for room phones to have single-digit access to hotel services and a multiple-digit access to room phones.

The Single-digit Access to Hotel Services feature allows a customer to define a pause timer, called a second-digit timer, between the first and second dialed digits, and allows two speed-call entries to be defined for a station group. The first speed-call entry is used for normal pretranslation. The second speed-call list is used when the second digit timer times out (that is, when time out occurs after the first digit is dialed, with the first digit in the first speed-call list being translated).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Dependency:

- Pretranslation (PXLT) package 92

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Single-digit Access to Hotel Services for each customer.

2 LD 18 – Define the Translation tables required by this feature.

LD 15 – Enable Single-digit Access to Hotel Services for each customer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and Options.
...		
- OPT	(SDDE) SDAL	(Deny) allow Single Digit Access.

LD 18 – Define the Translation tables required by this feature.

Prompt	Response	Description
REQ	CHG	Change
TYPE	PRE	Pretranslation calling group data block
CUST	xx	Customer number, as defined in LD 15
XLAT	xxx yyyy	Calling group number to translation Speed Call list number correlation. Format if International Supplementary Features (SUPP) package 131 is not equipped Where: • xxx = Pretranslation group number, 0-254 • xxx = Group 0 is used for trunks • xxx = Group 1 is used for attendant consoles. • xxx = Groups 2-254 can be used for other calling groups. • yyyy = List number to be used for Pretranslation, 0-8191. 8191 is used to remove the group from Pretranslation. Pretranslation group number. Format if international Supplementary Features (SUPP) package 131 is equipped Where: • xxx = Group 0 is used for trunks • xxx = Group 1 is used for attendant consoles. • xxx = Groups 2-254 can be used for other calling groups.
...		
- SDA	0-8190	Single-digit Access Speed Call List number

Feature operation

In the example that follows, if a room guest dials the digit 7, the guest's call is immediately terminated at DN 4300, the front desk. If the guest had dialed the digit 2, then after the second digit timer times out, the guest's call is terminated at DN 4002, laundry. If the guest enters three more digits (*xxx*) before the second digit time-out, the appropriate room number (2*xxx*) is rung.

Speed Call CodeDN Designation

0	Operator (00)
1	Room Service (4001)
2	Laundry (4002)
3	Concierge (4100)
4	Restaurant (4101)
5	Health Club (4200)
6	Maid (4201)
7	Front Desk (4300)
8	Toll Calls (88)
9	Local Calls (99)

First Entry Speed Call List (for normal pretranslation)

First Dialed DigitAction

1	Pass as 1
2	Pass as 2
3	Pass as 3
4	4101
5	4200
6	4201
7	4300
8	88
9	99
0	Pass as 0

Second Entry Speed Call List (for pretranslation after time out)

First Dialed Digit	Action
1	4000
2	4002
3	4100
4	N/A
5	N/A
6	N/A
7	N/A
8	N/A
9	N/A
0	0017

Slow Answer Recall Enhancement

Contents

This section contains information on the following topics:

Feature description 709
Operating parameters 711
Feature interactions 711
Feature packaging 712
Feature implementation 712
Feature operation 713

Feature description

This enhancement to the Slow Answer Recall feature changes how the recall is treated once presented to the Attendant Console. This enhancement applies to Integrated Services Digital Network (ISDN) and standalone environments.

If an incoming call extended by the attendant to a set is not answered after a preprogrammed time period, it is recalled to the Attendant Console. The call type may be indicated by an Incoming Call Indicator (ICI) key programmed to flash for recalls. The target set will continue to ring after the call is presented to the attendant. The target set can answer the call before the attendant does, in which case the call is cleared from the Attendant Console and the incoming call and target set will be connected.

If the attendant answers the recall before the target set, a speech connection is established between the calling party on the source (SRC) side of the console. The target set continues to ring while still being connected to the destination (DEST) side of the console. This feature only affects the operation after the attendant has answered the recall.

When a Slow Answer Recall occurs, the call is placed in the attendant queue and appears on the console. The target set will continue to ring while the recall is queued and presented on the console, but is unanswered. When the attendant answers the recall, by pressing the appropriate Loop key or the Recall ICI key, the target set will be disconnected as soon as the Attendant Console answers the Slow Answer Recall.

In a ISDN environment, the feature works in a similar way regardless of the location of the called party (on the same node as the attendant or on a remote node), and if Network Attendant Service (NAS) routing is involved in the call or not.

Call Waiting Recalls and Camp-on Recalls

This enhancement adds Call Waiting Recall and Camp-on Recall functionality to Slow Answer Recall. This enhancement applies within standalone and networking environments.

Call Waiting Recall

Within a standalone environment, if an incoming call extended by the attendant or a set (equipped with the Multi-Party Operations feature) to a busy station (equipped with Call Waiting) is not answered within a customer-defined period of time, it is recalled to the attendant. The recall is presented to the attendant or placed in the attendant queue.

Camp-on Recall

An incoming call is extended by the attendant or a set (equipped with the Station Camp-on feature) to a busy station that is not equipped with Call Waiting. The attendant or set camps -on the call to the target set. If the call is not answered within a customer-defined period of time, it is recalled to the attendant. The call is presented to the attendant or placed in the attendant queue. Until the attendant answers the call, the call remains camped-on to the target set and can still be answered. If the attendant answers the recall by pressing the appropriate loop key or the Recall key, the target set is disconnected and can no longer answer the call. The target set must be redialed to extend the call.

Within a network environment, the Call Waiting Recall and Camp-on Waiting Recall enhancements must be configured at a node. Both the Call Waiting Recall and Camp-on Waiting Recall enhancements operate in the same way as in the stand-alone case. The location of the calling and called party and the attendant have no affect on the call processing.

Network Attendant Service (NAS) is not required, but it may be applied at a node. In this case, NAS takes precedence over the Call Waiting Recall and Camp-on Waiting Recall Enhancements, in that the target set is disconnected from the call due to time-out and not to the attendant pressing the loop key or Recall key.

Operating parameters

The same as for Slow Answer Recall.

Feature interactions

Attendant Recall with Splitting
Multi-Party Operations
Secrecy Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Call Waiting Recall
Camp-on Waiting Recall

The Call Waiting Recall and Camp-on Waiting Recall enhancements are compatible with Station Camp-on (STCA).

A forced Camp-on override recall occurs to the attendant. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are equipped, the destination is automatically disconnected when the attendant answers. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are not equipped, and the attendant answers the recall at the same time that the destination answers, a conference is established between the attendant, source, and destination.

Intercept Computer Dial from Directory

If the attendant extends an SRC party to a DEST party on the local node, but slow answer recall occurs since the DEST does not answer, it is possible to dial a new DN from the ICP (the DEST is disconnected when the attendant answers).

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Enable Slow Answer Recall Enhancement for the customer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and Options data block
...		
- OPT	(SLD) SLA	Slow Answer Recall Enhancement (denied) allowed.

Feature operation

Slow Answer Recall Enhancement

When a Slow Answer Recall occurs the call is placed in the attendant queue and appears on the console. The target set will continue to ring while the recall is queued and presented on the console but unanswered. When the attendant answers the recall, by pressing the appropriate **Loop** key or the **Recall ICI** key, the target set will be disconnected as soon as the Attendant Console answers the Slow Answer Recall.

Call Waiting Recall

Until the attendant answers the call, it remains waiting on the target set, and can still be answered. If the attendant answers the recall by pressing the appropriate **Loop** key or the **Recall** key, the target set is disconnected and can no longer answer the call – the target set will have to be redialed to extend the call.

Camp-on Recall

Until the attendant answers the call, the call remains camped-on to the target set, and can still be answered. If the attendant answers the recall by pressing the appropriate Loop key or the Recall key, the target set is disconnected and can no longer answer the call; the target set will have to be redialed to extend the call.

Slow Answer Recall for Transferred External Trunks

Contents

This section contains information on the following topics:

Feature description 715
Operating parameters 716
Feature interactions 716
Feature packaging 717
Feature implementation 717
Feature operation 718

Feature description

This feature allows an external call to be transferred to a ringing set anywhere within an Integrated Services Digital Network (ISDN) network. The transferred call may be incoming or outgoing, supervised or unsupervised. If the call is not answered within a customer-defined period of time, it is routed to the local attendant as a slow answer recall.

Within a standalone environment, this capability is provided by the Multi-Party Operation feature.

An external call is a call originated by the Public Switched Telephone Network (PSTN). This includes calls originating on a Central Office (CO), Foreign Exchange (FEX), Direct Inward Dialing (DID), or Wide Area Telephone Service (WATS) trunk on a local or remote node, and calls from the PSTN to an ISDN node using Network Attendant Service (NAS) signaling protocol over an ISDN TIE trunk.

Operating parameters

This feature applies only to systems using Meridian Customer Defined Networking (MCDN) signaling over ISDN Signaling Link (ISL)/ISDN TIE links.

All network nodes must be configured with Network Attendant Service (NAS).

Feature interactions

AC15 Recall: Transfer from Norstar

In both standalone and Network Attendant Service (NAS) environments, when a call is transferred to a ringing set on the system by an AC15 trunk, the RTIM recall timer is not started.

Attendant Recall

Slow Answer Recall Modification (SLAM) has an interaction after the attendant answers the recall. If SLAM is configured, the target set is disconnected after the attendant answers the recall. If SLAM is not configured, the target set rings until the attendant releases it.

Call Forward No Answer

If the ringing station to which the call has been transferred has Call Forward No Answer active, the call will be transferred to the call forward DN after the specified number of ring cycles.

ICP Network Screen Activation, Flexible DN, Meridian Mail Interactions

When an Intercept Computer (ICP) position set transfers an external call across an ISDN network, the slow answer recall timer is set at the transferring node to prevent the terminating set to be rung indefinitely. When the slow answer recall timer times out, the transferred call is recalled to the attendant at the transferring node.

Multi-Party Operations

The Multiple Party Operation recall can only be applied in a standalone environment, and therefore does not interact with this feature.

Network Attendant Service Anti-tromboning

NAS Anti-tromboning is supported by this feature.

Feature packaging

- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145, **or**
- ISDN Signaling Link (ISL) package 147
- Network Attendant Service (NAS) package 159.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Enable Slow Answer Recall Enhancement for the customer.

2 LD 15 – Configure Timers for Slow Answer Recall for Transferred External Trunks.

LD 15 – Enable Slow Answer Recall Enhancement for the customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and Options data block
...		
- OPT	(SLD) SLA	Slow Answer Recall Enhancement (denied) allowed

LD 15 – Configure Timers for Slow Answer Recall for Transferred External Trunks.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	TIM	Timers
...		
- RTIM	xxx yyy zzz	Recall timers for Slow Answer, Camp-on and Call Waiting, where: xxx = 0-(30)-378 for Slow Answer yyy = 0-(30)-510 for Camp-on, and zzz = 0-(30)-510 for Call Waiting. These timers indicate in seconds the elapsed time before attendant recall. Slow Answer must be a multiple of six seconds. To change one timer, all three fields must be input.

Feature operation

No specific operating procedures are required to use this feature.

Source Included when Attendant Dials

Contents

This section contains information on the following topics:

Feature description 719
Operating parameters 720
Feature interactions 720
Feature packaging 722
Feature implementation 722
Feature operation 722

Feature description

This feature provides a new option in LD 15, which allows the customer to define whether or not the source is to be included in a call while the attendant is dialing the destination (SIAA = allow, SIAD = deny). If the destination answers while the attendant is still included in the call, intrusion tone is provided to all parties to indicate that a conference has been established. The intrusion tone is defined in LD 56, and is a prerequisite for the Source Included when Attendant Dials feature.

If SIAA has been defined, the source will be included in all situations, regardless of the state of the destination, except when the attendant is performing Break-In to a busy station.

The following table outlines the operation, if SIAA has been defined, according to the state of the destination party:

Destination	Source	
	Included	**Excluded**
Idle extension	x	
First Degree Busy	x	
Second Degree Busy	x	
Camp-on	x	
Intercept forwarded	x	
Line lock-out	x	
Vacant	x	
Busy, Attendant Break-in		x
Meridian Mail	x	
Recorded Announcement	x	
Music	x	

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will follow the current Source Included when Attendant Dialing handling occurs.

Attendant Break-In

The operation of the Break-In feature is not affected, except that the source receives busy tone before the attendant presses the Break-In (BKI) key.

Attendant Supervisory Console

While the attendant dials the destination, the source receives intrusion tone.

Automatic Call Distribution

The source is included in a conference involving the attendant, the source, and Automatic Call Distribution (ACD). When the call is answered by the ACD agent, intrusion tone is provided to all parties in the conference.

Camp-On
Semi-automatic Camp-On

The source remains included while the attendant dials the destination.

Intercept treatment

If the attendant dials a destination which is intercepted, the source remains included in the call.

Meridian Mail

The source is included in a conference involving the attendant, the source, and Meridian Mail answering. When the call is answered by Meridian Mail, the attendant and source receive intrusion tone.

Recorded Announcement
Music

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Secrecy Enhancement

Source Included when Attendant Dials takes precedence over Secrecy and Enhanced Secrecy.

Feature packaging

This feature requires:

- International Supplementary Features (SUPP) package 131
- Flexible Tone and Cadences (FTC) package 125
- Trunk Barring (TBAR) package 132

Feature implementation

LD 15 – Configure Source Included when Attendant Dials for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	FTR	Features and Options
...		
- OPT	(SIAD) SIAA	(Deny) or allow Source Included when Attendant Dials.

Feature operation

No specific operating procedures are required to use this feature.

Special Dial Tones after Dialed Numbers

Contents

This section contains information on the following topics:

Feature description 723
Operating parameters 724
Feature interactions 724
Feature packaging 725
Feature implementation 725
Feature operation 726

Feature description

This feature allows special dial tones to be provided after certain telephone numbers are dialed. Both the telephone numbers and associated dial tones are customer-defined in LD 56. The system can handle a list of up to 20 telephone numbers with a maximum length of five digits. A tone can be associated with each number. Several different tones can be provided during a dialing sequence by defining a tone with any combination of digits in the dialed number.

For example, for the number 12345, a tone can be provided after the digit 1 is dialed, after the digits 123 are dialed, and after the whole (12345) number is dialed. This is done by defining a tone with the digit 1, a tone with the digits 123, and a tone with the digits 12345.

When a number is dialed, the system performs digit analysis. As soon as the dialing sequence is recognized as part of the customer-defined list, the system provides the associated tone, if one has been defined. The tone is generated after all other treatment of digits is performed. As soon as another digit is dialed, the tone is removed. This digit analysis is done until the dialing sequence is completed.

Tones are provided to the following originating terminals:

- all types of sets (including data terminals) and attendants, and
- TIE trunks, except those with MFC/MFE signaling.

Operating parameters

The system performs digit analysis before any other treatment of digits, except digit insertion for incoming trunk calls.

In a network environment, digit recognition is reported to the distant node, which must be equipped to handle the processing.

Feature interactions

Digital Private Network Signaling System (DPNSS1)/Digital Access Signaling System (DASS2) Uniform Dialing Plan (UDP) Interworking

The Special Dial Tones after Dialed Numbers feature is supported in a DPNSS1 UDP network.

Digital Trunk Interface (DTI) – Commonwealth of Independent States (CIS)

Special Dial Tones can be used to provide dial tone after the system user has dialed the digit “9” (Local Exchange access code).

EuroISDN Master Mode

This feature is not supported for incoming calls on the ETSI network side, but it is supported for outgoing calls.

Special dial tone after access codes

Special dial tone after access codes takes precedence over the special dial tones after dialed number treatment. To define special dial tones after access codes, NO has to be entered in response to prompt DLTN in LD 86 (to inhibit dial tone to access codes). The access code digits and associated tones would then have to be defined in response to the DTAD prompt in LD 56.

Feature packaging

Flexible Numbering Plan (PNP) package 160; and to define SRC1-SRC8 special tones, the Flexible Tones and Cadences (FTC) package 125.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 86 – Enable Special Dial Tones after Dialed Numbers.

2 LD 15 – Configure Special Dial Tones after Dialed Numbers.

LD 86 – Enable Special Dial Tones after Dialed Numbers.

Prompt	Response	Comment
REQ	CHG	Change
CUST	xx	Customer number, as defined in LD 15
FEAT	ESN	ESN (Electronic switched network)
DLTN	(YES) NO	NARS/BARS Dial Tone after dialing AC1 or AC2 access codes.

LD 15 – Configure Special Dial Tones after Dialed Numbers.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	DTAD	Special Dial Tone after Dialed Number data block.
DDGT ...	x...x	Dialed digits (1-5 digits).
- TONE	a...a	Tone to be provided after the dialed digits Where a...a: (DIAL) = Dial Tone SPDT = Special Dial Tone SRC!-SRC8 = Source tones 1-8 (Valid if FTC package 125 is equipped)

Feature operation

No specific operating procedures are required to use this feature.

Special Signaling Protocols

Contents

This section contains information on the following topics:

Feature description 727
Operating parameters 728
Feature interactions 728
Feature packaging 728
Feature implementation 728
Feature operation 729

Feature description

This feature allows the existing Swedish analog (500/2500 type) telephones to be connected through analog TIE trunks to the system. These TIE trunks use Swedish signaling protocols. The TIE trunks can be divided into the following types:

- automatic
- semi-automatic
- tone, or
- Automatic Telephony (ATL) (when the Swedish ATL trunk support feature is equipped).

Operating parameters

The Swedish TIE trunk types do not apply to digital TIE trunks.

The Swedish TIE trunk types cannot be mixed on a route.

The Swedish TIE trunks require trunk cards of type TPC71 or TPC237. The trunk cards must be placed on specific Televerket (TVT) loops.

A semi-automatic or tone TIE trunk should not be connected to another system trunk. An incoming Public Exchange/Central Office trunk can be connected to an outgoing automatic TIE trunk.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 16 – Configure Trunk route for Special Signaling Protocols.

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route data block
CUST	xx	Customer number, as defined in LD 15
...		
TKTP	TIE SEMI TIE AUTO TIE TONE	Semi-automatic TIE trunk data block. Automatic TIE trunk data block. Tone TIE trunk data block.

Feature operation

No specific operating procedures are required to use this feature.

Special Trunk Support

Contents

This section contains information on the following topics:

Feature description 731
Operating parameters 731
Feature interactions 732
Feature packaging 732
Feature implementation 732
Feature operation 735

Feature description

This feature allows the interface of the system with the Swedish Automatic Telephony (ATL) military radio-link network.

Operating parameters

ATL trunks must never be used for tandem switching or for networks using Electronic Switched Network (ESN) proprietary signaling.

Echo suppression and loss adjustment cannot be effected through software change.

Modified TPC237 cards must be used for ATL trunks, and must be configured on loops specifically defined for Televerket (TVT) use. An SSO adapter is used between the ATL network trunk and the TPC237 card.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Configure Trunks for Special Trunk Support.

2 LD 16 – Enable Trunk Routes for Special Trunk Support.

LD 14 – Configure Trunks for Special Trunk Support.

Prompt	Response	Comment
REQ	CHG	Change
...		
TYPE	TIE	TIE Trunk data block.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
CUST	xx	Customer number, as defined in LD 15
...		
NCOS	(0)	Network Class of Service.

RTMB	 0-511 1-510 0-127 1-510	Route number and Member number For Large Systems For Small Systems and Succession 1000 systems
...		
MNDN	9	Manual Directory Number.
TGAR	(0)	Trunk Group Access Restriction.
SIGL	EAM	Trunk signaling. E&M two-wire.
...		
STRI	WNK	Wink or Fast Flash.
STRO	WNK	Wink or Fast Flash
SUPN	YES	Answer and disconnect supervision required.

LD 16 – Enable Trunk Routes for Special Trunk Support.

Prompt	**Response**	**Comment**
REQ	CHG	Change
...		
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
...		
ROUT	 0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE ATL	The ATL data block for Sweden.
...		
ICOG	IAO	Incoming and outgoing trunk.

...		
SRCH	RRB	Round Robin Hunting for outgoing trunk (start with the next lower trunk than the one seized).
...		
ACOD	xxxx	Access Code for the trunk route. The ACOD must not conflict with the numbering plan.
...		
CNTL	YES	Change controls or timers.
- TIMR	ODT 8064	End of dial tone for Digitone trunks in milliseconds.
- TIMR	EOD 8064	End of Dial, non-Digitone trunks in milliseconds.
- TIMR	DSI 20096	Disconnect Supervision in milliseconds.
- TIMR	ICF 896	Incoming Flash in milliseconds.
- TIMR	OGF 896	Outgoing Flash in milliseconds.
- TIMR	GTI 1152	Incoming Guard in milliseconds.
- TIMR	GTO 1152	Outgoing Guard in milliseconds.
- TIMR	OBA 120	Outgoing B-Answer. Time in seconds to wait for B-Answer on outgoing ATL trunks for Sweden.
- SST	4	Seizure Supervision Timer, in seconds.
NEDC	ETH	Either end control.
FEDC	ETH	Far End Disconnect Control. Either end.
...		
PANS	YES	Pseudo Answer can be sent on some types of trunks as soon as end of dialing is detected. SUPN in LD 14 should be YES, or PANS = YES has no meaning.

Feature operation

No specific operating procedures are required to use this feature.

Speed Call Delimiter

Contents

This section contains information on the following topics:

Feature description 737
Operating parameters 738
Feature interactions 738
Feature packaging 739
Feature implementation 739
Feature operation 740

Feature description

The Speed Call Delimiter feature meets the Chinese Ministry of Posts and Telecommunications requirements for the operation of Speed Call and System Speed Call. This feature operates similar to the Speed Call and System Speed Call with the exception of delimiters and confirmation tones.

The Speed Call Delimiter feature requires a Speed Call controller to enter an asterisk (*) between abbreviated numbers and telephone numbers when configuring speed call lists. An additional octothorpe (#) delimiter is required for Analog (2500-type) sets to indicate the end of dialing. If an octothorpe (#) is not entered, then the telephone number is not stored and the entry is not valid.

The octothorpe (#) delimiter has the flexibility of being programmed as mandatory or optional. The delimiter can be modified to something other than an octothorpe (#).

Operating parameters

An asterisk (*) delimiter is used when programming speed call lists only. An asterisk (*) can also be used as a three second delay.

No changes occur when a user wants to display a number stored against a list entry number. To display a stored entry the user presses the Display key and the Speed Call key and dials the list number. The list number cannot be abbreviated.

This feature does not apply to Analog 500-type telephones.

The use of confirmation tone or announcement implies the use of an (#) as end of dial speed call delimiter. This means that an (#) cannot be stored as part of the digit string.

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network access code and dialed number are part of the Speed Call or Autodial Key. If the (#) is not entered, then the user receives a fast busy tone. Therefore if MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe at the FFCS prompt in LD 15.

Feature interactions

Autodial
Speed Call

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Group Call List

Speed Call Delimiter does not interact with Group Call List.

Outpulsing Asterisk (*) and Octothorpe (#)

If the Outpulsing Asterisk (*) and Octothorpe (#) (OPAO) package 104 is equipped and the configuration tone is programmed, then the value stored in the STRG prompt (LD 15) is entered rather than an octothorpe (#) to indicate the end of dial string. Following this, the numbers are stored.

Feature packaging

China Speed Call Delimiter requires Speed Call (OBTF) package 1 and System Speed Call (SSC) package 34.

Flexible Feature Codes (FFC) package 139 is required for Analog 2500-type telephones, if a set accesses speed call list or system speed call list or attendant console. This package is optional for proprietary sets or Attendant Consoles because these sets can access Speed Call List/System Speed Call List by using a key.

Feature implementation

To enable Speed Call and System Speed Call, the maximum number of speed call lists must be determined in LD 17. The speed call list memory size must also be configured in LD 18. For more information on these overlays and the assignment of these features to proprietary, analog (500/2500-type) telephones and attendant consoles please refer to the sections entitled Speed Call and System Speed Call in this publication.

LD 15 – Enable Speed Call Delimiter in Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Features and options
...		
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
- LEND	YES	List Entry Number Delimiter. If LEND=YES, then an asterisk (*) delimiter between the list entry number and telephone number must be entered. If LEND=NO, then existing Speed Call operation continues.
- MSCD	YES	Mandatory Speed Call Delimiter. Default = Octothorpe (#). An octothorpe (#) is required after entering telephone number to indicate the end of dial. If MSCD=NO, then the end of dial Speed Call Delimiter octothorpe (#) is optional.

Note: The China market requires an octothorpe (#) delimiter at the end of dialing. Other markets have the option of selecting a mandatory or optional delimiter by entering "YES" or "NO" at the MSCD prompt. The end of dial delimiter can be an octothorpe (default value) or it can be changed to another delimiter by modifying values at the Flexible Feature Code end-of-dialing indicator (FFCS). String to indicate end-of-dialing (STRG) and string length of end-of-dial indictor (STRL) prompts in LD 15.

Feature operation

Speed Call Delimiter Operation

Analog 2500-type telephone

- To Program Speed Call List - Go off-hook, dial and receive dial tone. Dial System Speed Call Controller (SCC) FFC code, the list entry number and telephone number (for example, *51*1*5556667777#). Get response. If accepted, then confirmation tone or announcement is

configured and the end of dial speed call delimiter is entered. Response is a tone or speech signal. Otherwise, silence is given. Go on-hook.

- To Use - Go off-hook and receive dial tone. Dial Speed Call User (SCU) code and the list entry number.
- To Delete List Entry Number in Speed Call List - Go off-hook and receive dial tone. Dial Speed Call Erase (SCE) FFC followed by list entry number (0 - 999) and (#) delimiter. Delete the specific list entry number.

Proprietary telephones and Attendant Console

- To Program Speed Call List – Press Speed Call Controller Key and the indicator flashes. Dial list entry number (0 - 9999) followed by an asterisk (for example, 1*5556667777). Press Speed Call Controller Key again. If entry is accepted, the indicator goes off. If the entry is not accepted, then the indicator remains flashing. An asterisk is only used to indicate the end of dial of list entry number and is not stored as a digit string.
- To Use on Proprietary Telephones – Lift the handset or press DN Key. Press System Speed Call Controller (SCC) or Speed Call User (SCU) Key. Dial the list entry number.
- Use Attendant Console – Press an idle loop key and then press Speed Call Controller (SCC) Key. Dial the list entry number.

System Speed Call Delimiter Operation

Proprietary telephones

- To Program System Speed Call List – Press assigned System Speed Call Controller Key and indicator flashes. Dial list entry number (0 - 999) followed by an asterisk (*) and then the telephone number (for example, 1*0115556667777). Then press SSC/SSU Key again. If accepted, the indictor goes off. If not accepted, the indicator remains flashing.
- To Use - Lift handset or press DN key. Press SSC/SSU Key. Dial the list entry number, or lift handset or press DN key. Dial SSU FFC code. Dial list entry number.

Attendant Console

- To Program System Speed Call List - Press SSC Key and indicator flashes. Dial list entry number (0 - 999), followed by an asterisk (*) and the telephone number. Press SSC Key again. If entry is accepted, indicator goes off. If the entry is not accepted, the indicator continues to flash.
- To Use - Press an idle loop Key. Dial SSU FFC code. Dial list entry number.

Speed Call Directory Number Access

Contents

This section contains information on the following topics:

Feature description 743
Operating parameters 744
Feature interactions 744
Feature packaging 744
Feature implementation 745
Feature operation 746

Feature description

The Speed Call Directory Number (DN) Access feature allows a Pilot DN to be used as an access code to either a Speed Call List (SCL) or a System Speed Call List (SSC).

Speed Call DN Access provides an alternative way to access either a Speed Call List or a System Speed Call List. Instead of dialing the Special Prefix (SPRE), a SCL or SSC access code, and a list entry number, or instead of depressing an idle DN key, a SCL or SSC key, and then dialing a list entry number, a user can alternatively dial a speed call access Pilot DN followed by the list entry number.

Since each speed call access Pilot DN is associated with a SCL or SSC list, users can access as many SCL or SSC lists as they need by dialing the appropriate Pilot DN.

A Pilot DN can be accessed from anywhere in a network, so that any network user can access all speed call lists defined for a network, from anywhere in the network. This allows a centralized Speed Call List to be set up for the entire network.

Operating parameters

The requirements for Speed Call and System Speed Call also apply to this feature.

Feature interactions

Direct Inward Dialing (DID) and TIE trunk access

An additional one to three digits will be accepted from these trunks to complete a Speed Call, provided these additional digits are allowed to be sent by the external system.

Speed Call
System Speed Call

Speed Call DN Access is an enhancement of the SCL and SSC features. Refer to SCL and SSC feature descriptions for interactions with other features.

Feature packaging

Speed Call Directory Number Access requires Group Hunt/DN Access to SCL (PLDN) package 120.

Dependencies:

- International Supplementary Features (SUPP) package 131
- Flexible Feature Codes (FFC) package 139
- System Speed Call (SSC) package 34
- Optional Features (OPTF) number 1

Feature implementation

LD 57 – Define, change, print, or remove data associated with FFC. A new PLDN prompt is introduced for Pilot DNs. The new LSNO prompt is used to associate the Pilot DN with a SCL or SSC list. The USE prompt is displayed only if the Pilot DN entered in response to the PLDN prompt has not already been defined.

Prompt	Response	Description
REQ	CHG NEW	Modify or create data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Code to be modified or created: Pilot DN.
PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be modified or created; enter carriage return to proceed to next prompt.
USE	 SCLC SCLU	USE: enter USE for Pilot DN. Speed Call List Controller. Speed Call List User.
LSNO	xxxx	List Number: enter Speed Call or System Speed Call list number. Speed Call list must exist in LD 18.

Prompt	Response	Description
REQ	OUT PRT	Remove or print a code or data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
CODE	PLDN ALL	Code requested: Pilot DN. All FFC.
PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be removed enter carriage return to proceed next prompt

Feature operation

To access either a Speed Call List or a System Speed Call List using this feature, dial a speed call access Pilot DN followed by the list entry number.

Pilot DN

Pilot DNs are defined as PLDN Flexible Feature Codes (FFC) via service change LD 57.

Pilot DNs can be used in two ways:

1 If the USE prompt is set to GPHT, the Pilot DN is defined to activate Group Hunting.

2 If the USE prompt is set to SCLC (Speed Call List Controller) or SCLU (Speed Call List User), the Pilot DN is defined to access the Speed Call or System Speed Call lists that are associated with the Pilot DN.

When the response to the USE prompt is SCLC (controller), a station can modify an SCL or SSC list by dialing the speed call access Pilot DN associated with that list, followed by a one- to three-digit list entry number, the number to be entered in the list, and then going on-hook.

Overflow tone is returned if the information entered is not valid. Confirmation tone is returned if the Flexible Feature Confirmation Tone (FFCT) option is set and trailing '#' is dialed, as in existing Flexible Feature Codes (FFCs) operations.

When the response to the USE prompt is SCLU (user), to use any entry in a SCL or SSC list, a station user dials the speed call access Pilot DN associated with the list, followed by the one- to three-digit list entry number.

Speed Call on Private Lines

Contents

This section contains information on the following topics:

Feature description 747
Operating parameters 747
Feature interactions 748
Feature packaging 748
Feature implementation 748
Feature operation 748

Feature description

This feature allows Meridian 1 proprietary telephone users equipped with a Private Line (PVR or PVN) key and a Speed Call (SCL) key to first access a Private Line trunk (by pressing the PVR or PVN key) and then make a speed call (by pressing the SCL key).

Operating parameters

When a Private Line call is made, recognizable Route Access Codes are absorbed from the start of every entry in the Speed Call List (for example, if 7654 is stored as a Speed Call List entry, and 76 is a valid Route Access Code, 76 is absorbed and 54 is outpulsed).

Feature interactions

Automatic Redial
Private Line
Speed Call features

The Automatic Redial (ARDL) feature is activated on a number dialed using the Private Line (PVR/PVN) key and then making a speed call by pressing the Speed Call (SCL) key.

Basic/Network Alternate Route Selection (BARS/NARS)

The BARS and NARS access codes (AC1 and AC2) are not absorbed. If a user has a Speed Call list entry that includes either AC1 or AC2, this entry will not terminate correctly when used on a Private Line. The BARS or NARS access code (AC1 or AC2) will be outpulsed, causing the Public Network to either terminate the call at an unwanted location or reject the call.

Feature packaging

Speed Call on Private Lines is part of base system software and requires Optional Features (OPTF) package 1.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

	ACTION	RESPONSE
1	User presses Private Line Ringing (**PVR**) or Private Line Nonringing (**PVN**) key.	Trunk is accessed and dial tone is returned.
2	User presses Speed Call key and enters list entry number.	The number stored against this entry is outpulsed.

Speed Call

Contents

This section contains information on the following topics:

Feature description 749
Operating parameters 750
Feature interactions 750
Feature packaging 754
Feature implementation 754
Feature operation 758

Feature description

Speed Call allows you to place calls by dialing a one-, two-, or three-digit code. You can use Speed Call for both internal and external calls. To use Speed Call, Meridian 1 proprietary telephones, and Attendant Consoles can have a Speed Call key/lamp pair.

Analog (500/2500 type) telephones can activate Speed Call by using Special Prefix (SPRE) or Flexible Feature Codes (FFC).

Analog (500/2500 type) telephones, Meridian 1 proprietary telephones, and Attendant Consoles can be designated as a Speed Call Controller (SCC) or a Speed Call User (SCU). SCCs can program the numbers to be stored (Speed Call codes) and can use the Speed Call list. SPUs cannot program Speed Call codes; they can only use the Speed Call lists.

Each stored number is assigned a Speed Call code from the Speed Call list. Each list can contain up to 1000 telephone numbers (entries). The maximum number of digits of the telephone number that can be stored in each entry is specified by the customer. Speed Call entries can be 4, 8, 12, 16, 20, 24, 28, or 31 digits long.

Operating parameters

You can define up to 8191 (0-8190) Speed Call lists per system, as long as sufficient memory is available. The maximum includes all combined Speed Call, System Speed Call (SSC), and Hot Line lists.

You can have as many Speed Call lists as you have available key/lamp pairs on any Meridian 1 proprietary telephone, or Attendant Console. Any number of users can be assigned to a list. Analog (500/2500 type) telephones can access only one Speed Call list. More than one Speed Call Controller can be assigned to each list, but this is not recommended.

A maximum of 31 digits for the telephone number is allowed per Speed Call list entry. An asterisk (*), which indicates a pause, and an octothorpe (#), which indicates end-of-dialing, can be programmed as part of the entry.

Speed Call list entries can be defined in LD 18 or by Speed Call Controllers. Speed Call Controllers must know the digit length (one, two, or three) required for the Speed Call codes in each list.

Feature interactions

AC15 Recall: Transfer from Meridian 1

Speed Call and Network Speed Call are supported with the AC15 Recall: Transfer from Meridian 1 on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone is detected by the system.

Additional transfers are supported if the digits are outpulsed without any treatment. For example, the route access code will be outpulsed to the far end. No dial tone detector is assigned and no timer is started so the digits are outpulsed immediately without checking the state at the far end.

Autodial Tandem Transfer

The Speed Call key cannot be used after a Centrex Switchhook Flash or during an established call to send digits out to the far site. The Speed Call key can be used only during the dialing stage.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using Speed Call (SCL) and System Speed Call (SSU/SSC) keys.

Call Forward/Hunt Override Via Flexible Feature Code

The Call Forward/Hunt Override FFC cannot be stored in a speed call list

Call Park

Speed Call can be programmed to parked calls or access parked calls.

Call Party Name Display

No name information displays during the programming of Speed Call numbers.

Calling Party Privacy

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Charge Account and Calling Party Number

Charge account numbers, including the Charge Account access Special Prefix (SPRE) code, can be stored as Speed Call or Autodial numbers. All current limitations of these features apply, such as a maximum of 23 digits per entry, including the access code. An Autodial number or dialed digits can follow, but not precede, a Speed Call number. The digits generated by an Autodial key during feature operation are accepted as Charge Account digits.

Charge Account, Forced

Forced Charge Account numbers (including the Special Prefix [SPRE] code and the Charge Account access code) can be entered in Speed Call lists or stored as Autodial numbers. The digits can also be stored, provided that the account number, regardless of its length, is followed directly by an octothorpe (#).

Enhanced Flexible Feature Codes - Outgoing Call Barring

Digits dialed using Speed Call are checked against the active Outgoing Call Barring (OCB) level. This includes calls made using the Dial Access to Speed Call feature (that is, using Pilot DNs).

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Speed Call.

Direct Private Network Access

If a Speed Call entry is programmed with a valid Authcode for Authcode Last followed by an octothorpe “#”, the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Last Number Redial

A number dialed using Speed Call will become the Last Number Redial number on all telephones, except the M2317 and M3000.

Pretranslation

A Speed Call List number should be programmed to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

If Pretranslation is enabled for a customer, then when a Speed Call List is assigned to a Pretranslation group within the customer, it cannot be accessed by a Meridian 1 proprietary set from within that customer group.

Scheduled Access Restrictions

The System Speed Call features ignore the Class of Service and TGAR access restrictions in a Scheduled Access Restriction schedule, using the Class of Service and NCOS defined in the speed call list.

Speed Call Delimiter

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Speed Call Directory Number Access

Speed Call DN Access is an enhancement of the Speed Call List (SCL) and System Speed Call (SSC) List features. Refer to “Speed Call, System” on page 759 for interactions with other features.

Station Specific Authorization Code

Station Specific Authorization Code (SSAU) feature treats stored autodial numbers as if they were entered at the telephone.

Three Wire Analog Trunk – Commonwealth of Independent States (CIS)

Speed Call on an E3W trunk will fail for toll calls. E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

User Selectable Call Redirection

Speed Call is not supported by User Selectable Call Redirection.

Feature packaging

Speed Call is part of Optional Features (OPTF) package 1, and has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Set maximum number of Speed Call lists.

2 LD 18 – Determine if there are enough memory and disk records for new Speed Call Lists.

3 LD 18 – Add a new Speed Call list.

4 LD 10 – Assign a Speed Call to an Analog (500/2500 type) telephone.

5 LD 11 – Assign a Speed Call list to proprietary telephone.

6 LD 12 – Assign a Speed Call list to an Attendant Console.

LD 17 – Set maximum number of Speed Call lists.

Prompt	Response	Description
REQ	CHG	Change.

TYPE	PARM	System Parameters Datablock
...		
- MSCL	0-8191	Maximum number of Speed Call lists.

LD 18 – Determine if there are enough memory and disk records for new Speed Call Lists.

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8191	Number of lists to be added.
DNSZ	4-(16)-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of entries in Speed Call list.
Note: Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.		

LD 18 – Add a new Speed Call list.

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove a Speed Call list.
TYPE	SCL	Speed Call data block.
LNSO	0-8190	Speed Call list number.
DNSZ	4-(16)-31	Maximum number of digits in a list entry (that is, 4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list may be updated.

STOR	xxx yy...yy	xxx = list entry number (0-9, 00-99, or 000-999). yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	(YES) NO	Data is correct and list can be updated.

Note: The prompt WRT follows prompts SIZE and STOR, asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating the data was not stored and must be reentered.

A response of **** aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved.

The following information is output with the WRT prompt, following SIZE:

ADDS: MEM: xxxxx DISK: yy.y

where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new Speed Call list. Check the MEM AVAIL and DISK REC AVAIL values output before the REQ prompt.

LD 10 – Assign a Speed Call to an Analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	500	Telephone type
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FTR	SCU yyyy SCC yyyy	Speed Call User, list number (0-8190) Speed Call Controller, list number (0-8190)

LD 11 – Assign a Speed Call list to proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx SCU yyyy xx SCC yyyy	System Speed Call User key Speed Call Controller key, where: xx = key number, and yyyy = Speed Call list number (0-8190) M3000 must use key 21. M2317 must use key 0-10 or key 21

LD 12 – Assign a Speed Call list to an Attendant Console.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ATT 1250 2250	Console type
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx SCC yyyy	Speed Call Controller, where: xx = key number, and yyyy = list number (0-8190)

Feature operation

To store Speed Call entries from a Meridian 1 proprietary telephone, or Attendant Console (Controller):

- Without lifting the handset, press **Speed Call.** The indicator flashes.
- Dial the Speed Call code (0-999), followed by the phone number it represents.
- Press **Speed Call.** If the entry is accepted, the indicator goes off. If the entry is not accepted, the indicator continues flashing.

To make a Speed Call from a Meridian 1 proprietary telephone, or Attendant Console (User):

- Lift the handset and press **Speed Call** (telephone).
  - — Select an idle loop key and press **Speed Call** (Attendant Console).
- Dial the Speed Call code. The telephone number represented by the Speed Call code is dialed automatically.

To store Speed Call entries from an analog (500/2500 type) telephone (Controller):

- Lift the handset and press octothorpe (#) +2 (2500 telephone) or SPRE+75 (analog (500/2500 type) telephone).
- Dial the Speed Call code (0-999), followed by the phone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.
- Hang up.

Repeat steps 1 through 3 for each entry to be stored.

To make a Speed Call from an analog (500/2500 type) telephone (User):

- Lift the handset and dial #3 (2500 telephone), or SPRE 76 (analog (500/2500 type) telephone).
- Dial the Speed Call code (0-999). The telephone number represented by the Speed Call code is dialed automatically.

  Note: In addition to SPRE codes your system may be equipped with Flexible Feature Codes (FFCs).

Speed Call, System

Contents

This section contains information on the following topics:

Feature description 759
Operating parameters 760
Feature interactions 760
Feature packaging 762
Feature implementation 763
Feature operation 767

Feature description

System Speed Call extends the capabilities of Speed Call. In addition to abbreviated dialing, System Speed Call allows a user to temporarily override the telephone's Class of Service, Trunk Group Access Restrictions (TGARs), and code restrictions.

Analog (500/2500 type) telephones, Meridian 1 proprietary telephones, and Attendant Consoles can activate System Speed Call by using SPRE or Flexible Feature Codes (FFC).

An analog (500/2500 type) telephone can be designated as a System Speed Call User only (not Controller) and can access one System Speed Call list. Meridian 1 proprietary telephones can be System Speed Call Users (SPRE codes or key access) or Controllers (key access only). Attendant Consoles can be System Speed Call Users (dial access only) and System Speed Call Controllers (key access only).

Operating parameters

Up to 8191 (0-8190) Speed Call lists are allowed as long as sufficient memory is available. The new maximum includes all combined Speed Call, System Speed Call and Hot Line lists, 4096 (0-4095) of which can be System Speed Call lists.

System Speed Call lists can have up to 1000 entries and each entry can be up to 31 digits in length.

Restrictions applied to a telephone are ignored only for the origination of a call made through System Speed Call. Restrictions are applied if any call modification is attempted once the call is established.

System Speed Call lists can only be programmed in LD 18 or from telephones or Attendant Consoles equipped with a System Speed Call Controller key.

The technician can add or copy up to 100 System and regular Speed Call Lists at a time.

Feature interactions

Attendant Administration

System Speed Call lists can be assigned using Attendant Administration.

Authorization Code Security Enhancement

If the Basic Authorization Code (BAUT) or Network Authorization Code (NAUT) package is equipped, a Network Class of Service (NCOS) is assigned to the System Speed Call list. The NCOS of the System Speed Call list replaces the NCOS of the Authorization code or Forced Charge Account code if it increases the Facility Restriction Level (FRL) of the code.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using System Speed Call (SSU/SSC).

Basic/Network Alternate Route Selection (BARS/NARS)

If the BARS or NARS package is equipped, an NCOS is assigned to the System Speed Call list. The NCOS associated with the System Speed Call list replaces the NCOS of the telephone if it increases the Facility Restriction Level (FRL) of the user.

Calling Party Privacy

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Capacity Expansion

Any number from 0 to 4095 can be assigned to a System Speed Call list.

China – Flexible Feature Codes - Outgoing Call Barring

Digits dialed using System Speed Call are checked against the active OCB level.

Flexible Feature Code

With Flexible Feature Code (FFC), a confirmation tone is provided for Speed Call store after the end-of-dial (EOD) string is entered.

Hot Line

When the System Speed Call package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

Last Number Redial

A number dialed using a System Speed Call key becomes the Last Number Redial number on all telephones, except the M2317 and M3000. A number dialed using SPRE-activated System Speed Call becomes the Last Number Redial number on all telephones. The original Class of Service and NCOS restrictions of the telephone apply when using Last Number Redial.

Off-Hook Alarm Security

Off-Hook Alarm Security (OHAS) treatment can apply to these features if the ASTM expires. The Alarm Security Timer may expire for the following reasons:

- A dial tone or interdigit timeout occurs while dialing the speed call access code.
- The Speed Call being accessed has an asterisk (*) causing a three-second delay. If the ASTM is three seconds or less, the OHAS intercept treatment may occur.

Pretranslation

Program a Speed Call List number to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

Feature packaging

System Speed Call (SSC) package 34 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Set maximum number of Speed Call lists.

2 LD 18 – Compute Speed Call list memory size and disk records. Use this prompt sequence to determine if there is enough memory and disk space for new Speed Call lists. Compare the output with the "MEM AVAIL" and "DISK AVAIL" values output before the REQ prompt.

3 LD 18 – Add or change a System Speed Call list.

4 LD 10 – Add or change System Speed Call for Analog (500/2500 type) telephones.

5 LD 11 – Add or change System Speed Call list for Meridian 1 proprietary telephones.

6 LD 12 – Add or change a System Speed Call list for Attendant Consoles.

7 LD 20 – Print Speed Call data. Respond to the TYPE prompt with SCL to print regular and System Speed Call lists and pretranslation. Respond to the TYPE prompt with SSL to print the System Speed Call data block.

LD 17 – Set maximum number of Speed Call lists.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System parameters
...		
- MSCL	0-8190	Maximum number of Speed Call lists.

LD 18 – Compute Speed Call list memory size and disk records. Use this prompt sequence to determine if there is enough memory and disk space for new Speed Call lists. Compare the output with the "MEM AVAIL" and "DISK AVAIL" values output before the REQ prompt.

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8190	Number of lists to be added.
DNSZ	4-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of entries in Speed Call list.

LD 18 – Add or change a System Speed Call list.

Prompt	Response	Description
REQ	NEW CHG OUT NEW xx, CPY xx	Add, change, or remove a single speed call list; Add or copy xx lists.
TYPE	SSC SCL	System Speed Call. Speed Call List.
LSNO	0-8190 xxxx yyyy	Number of list to add, where: xxxx = number of list to be copied, and yyyy = number of list to receive copy.
NCOS	0-99	NCOS to be assigned to calls accessing the list.
DNSZ	4-(16)-31	Maximum number of digits in a list entry (that is, 4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list may be updated.
STOR	xxx yy...yy	xxx = list entry number (0-9, 0-99, or 0-999). yy = digits to be stored against the entry (must be equal to or less than DNSZ).

WRT	(YES) NO	Data is correct and list may be updated.
Note: The prompt WRT follows prompts SIZE and STOR asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating the data was not stored and must be reentered. A response of "****" aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved. The following information is output with the WRT prompt, following SIZE: ADDS: MEM: xxxxx DISK: yy.y Where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new Speed Call list. Check the "MEM AVAIL" and "DISK REC AVAIL" values output before the REQ prompt.		

LD 10 – Add or change System Speed Call for Analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
FTR	SSU yyyy	System Speed Call user, list number (0-4095).

LD 11 – Add or change System Speed Call list for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

SSU	yyyy	System Speed Call list number (0-4095) for dial access.
KEY	xx SSU yyyy	System Speed Call user key.
	xx SSC yyyy	System Speed Call Controller key, where: xx = key number, and yyyy = System Speed Call list number (0-4095). ***Note:*** The M2317 and M3000 must use key 21.

LD 12 – Add or change a System Speed Call list for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SSU	yyyy	System Speed Call list number (0-4095) for dial access.
KEY	xx SSC yyyy	System Speed Call Controller key, where: xx = key number, and yyyy = System Speed Call list number (0-4095).

LD 20 – Print Speed Call data. Respond to the TYPE prompt with SCL to print regular and System Speed Call lists and pretranslation. Respond to the TYPE prompt with SSL to print the System Speed Call data block.

Prompt	Response	Description
REQ	PRT	Print.
TYPE	SCL	Regular and system speed call lists.
LSNO	0-8190	List number for speed call or system speed call print for all lists.

RNGE	xxxx xxxx	Range of all speed call entries (0-1000) to be printed. Print all entries

Feature operation

To store System Speed Call entries from a Meridian 1 proprietary telephone, or Attendant Console (Controller):

1 Without lifting the handset, press **Speed Call**. The indicator flashes.

2 Dial the Speed Call code (0-999), followed by the telephone number it represents.

3 Press **Speed Call**. If the entry is accepted, the indicator goes off. If the entry is not accepted, the indicator remains flashing.

To make a System Speed Call from a Meridian 1 proprietary telephone, or Attendant Console (User):

1 Lift the handset and dial SPRE 73 or press the System Speed Call key (telephone).

– or –

Select an idle loop key and dial SPRE 73 (Attendant Console).

2 Dial the Speed Call code.

If the Speed Call number is accepted, the telephone number represented by the Speed Call code is dialed automatically. No confirmation tone is given unless Flexible Feature Code (FFC) is implemented.

If the Speed Call number is not accepted, a fast busy signal indicates the number was rejected.

To make a System Speed Call from an analog (500/2500 type) telephone (User):

1 Lift the handset and dial SPRE 73.

2 Dial the Speed Call code (0-999). The telephone number represented by the Speed Call code is dialed automatically.

Note: In addition to SPRE codes your system can be equipped with Flexible Feature Codes.

The routine to add a call list aborts under the following conditions:

- trying to add a call list whose number is already in use, or
- trying to add multiple call lists when there is insufficient memory.

Speed Call/Autodial with Authorization Codes

Contents

This section contains information on the following topics:

Feature description 769
Operating parameters 770
Feature interactions 770
Feature packaging 770
Feature implementation 771
Feature operation 771

Feature description

This feature is an enhancement to the existing Speed Call and Autodial features. It allows a Speed Call entry to contain an Authorization Code with an associated trunk route or Electronic Switched Network (ESN) access code and dialed number. The digits stored are recorded in Call Detail Recording (CDR), if equipped, for billing purposes.

The Speed Call entry can be one of the following:

- SPRE + 6 + Authorization Code
- SPRE + 6 + Authorization Code + #, or
- SPRE + 6 + Authorization Code + # + ESN access code and dialed number.

Operating parameters

Authorization Code Conditionally Last is not supported.

An octothorpe (#) is required as a delimiter after the Authorization Code if an ESN access code and dialed number are stored as part of the Speed Call or Autodial key. If the octothorpe is not entered, the user receives a fast busy tone. The octothorpe is not stored in the CDR record.

If the system initializes before the Authorization Code is recorded by CDR, the record may be lost.

The M3000 set can display up to 29 digits. Additional digits cause the digits to scroll off the display. Only one softkey, key 21, can be programmed for Speed Call.

An M2317 set can display up to 31 digits.

For Meridian 1 proprietary telephones, up to 31 digits per Speed Call entry are allowed.

On digit display sets, Authorization Codes cannot be blocked from being displayed.

There is no validation of the Authorization Code until the Speed Call key is activated.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The following packages are required to implement this feature:

- Basic Authorization Code (BAUT) package 25, or Network Authorization Code (NAUT) package 63.
- Optional features (OPTF) package 1, System Speed Call (SSC) package 34, or Network Speed Call (NSC) package 39.

Feature implementation

An Authorization Code can be entered as part of a Speed Call list.

Feature operation

No specific operating procedures are required to use this feature.

Station Activity Records

Contents

This section contains information on the following topics:

Feature description 773
Operating parameters 773
Feature interactions 774
Feature packaging 775
Feature implementation 775
Feature operation 777

Feature description

When a set is configured with Class of Service Call Detail Monitoring Allowed (CDMA) for all incoming and outgoing calls, Station Activity Records are produced. The format of Station Activity Records is identical to other Call Detail Recording (CDR) records, but they have a new type of identifier (D). Existing CDR records are not affected by this new functionality.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Redirection

A Station Activity Record is only produced for a set designated as CDMA that is involved in a call with a trunk. A Station Activity Record is not generated for any set which does not answer the call, regardless of whether it has Class of Service CDMA or CDMD. Any other CDR records generated during call redirection are not affected.

Call Transfer

A Station Activity Record is generated when a set with Class of Service CDMA transfers a trunk call. CDR “X” record generation is not affected by this development. The set to which the call is transferred also produces a Station Activity Record if it has Class of Service CDMA and answers the call. When the second “D” record is produced (by the set to which the call is transferred), the digits field of the “D” record shows the digits dialed by the transferring set.

Conference

For a set with Class of Service CDMA involved in a call with a trunk, a Station Activity Record is produced only when that set conferences in the first party. Conferencing of all subsequent parties does not generate a “D” record. An additional “D” record is produced when the last conferee with Class of Service CDMA connected to the trunk goes on hook. This does not affect any other CDR record generation during a conference.

Internal Call Detail Recording

Internal Call Detail Recording records are produced according to the Class of Service ICDA/ICDD of a set. The Station Activity Record enhancement does not affect the ICDR record generation.

Feature packaging

Station Activity Records is package 251 (SCDR).

Dependencies:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Terminal (CTY) package 5

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Set Class of Service CDMA/CDMD for an analog (500/2500 type) telephone.

2 LD 11 – Set Class of Service CDMA/CDMD for Meridian 1 proprietary telephones.

3 LD 27 – Set Class of Service CDMA/CDMD for BRI sets.

4 LD 17 – Define a CDR link for Call Detail Recording.

5 LD 15 – CDR must be enabled for the customer.

LD 10 – Set Class of Service CDMA/CDMD for an analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	500	Analog (500/2500 type) telephone.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 11 – Set Class of Service CDMA/CDMD for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	xxxx	Meridian 1 proprietary telephone type.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 27 – Set Class of Service CDMA/CDMD for BRI sets.

Prompt	Response	Description
REQ	NEW CHG PRT	New, change, or print.
TYPE	DSL	Digital Subscriber Loop.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the set (when the trunk is involved in the call). CDMD denies record generation.

LD 17 – Define a CDR link for Call Detail Recording.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ADAN	All input / output devices (includes D channels).
USER	CTY	TTY has CTY as the user (for CDR records).

LD 15 – CDR must be enabled for the customer.

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	CDR	Call Detail Recording
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CDR	YES	Call Detail Recording.
- PORT	0-15	The CDR port number for the customer.

Feature operation

No specific operating procedures are required to use this feature.

Station Category Indication

Contents

This section contains information on the following topics:

Feature description 779
Operating parameters 779
Feature interactions 780
Feature packaging 780
Feature implementation 780
Feature operation 782

Feature description

The Station Category Indication (SCI) feature allows an attendant to selectively answer internal attendant Directory Number (DN) calls on a priority basis. Stations are assigned a category, with priority indicated by an Incoming Call Indicator (ICI) lamp at each Attendant Console. Using the answering priority defined in LD 15, the attendant gives prompt attention to a call presented at a high-priority ICI lamp by selecting the associated ICI key.

Operating parameters

A maximum of seven station categories (1-7) can be assigned.

Calls from SCI 0 stations appear on the dial 0 ICI.

Calls from fully restricted stations appear on the dial 0 fully restricted ICI.

The Station Category Indication (SCI) feature should not be mixed with any other Incoming Call Indicator (ICI) assignment on the same ICI key/lamp pair.

Feature interactions

Centralized Attendant Service

When Centralized Attendant Service (CAS) is active, calls from a remote station to the attendant DN appear on the remote ICI key/lamp pair at the CAS main, regardless of the station SCI category.

Controlled Class of Service

The Controlled Class of Service (CCOS) feature has priority over SCI. A station's SCI category is suppressed when CCOS is active, and calls to the attendant DN carry the CCOS class defined in the database.

Phantom Terminal Numbers (TNs)

SCI cannot be enabled on a Phantom TN.

Feature packaging

Station Category Indication (SCI) package 80 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Add or change a Station Category Indication ICI key/lamp pair for Attendant Consoles.

2 LD 10 – Change SCI for Analog (500/2500 type) telephones.

3 LD 11 – Change SCI for Meridian 1 proprietary telephones.

LD 15 – Add or change a Station Category Indication ICI key/lamp pair for Attendant Consoles.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CDB	Customer Data Block.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
ICI	0-19 CA1-CA7	Assign ICI key/lamp pair for SCI.
ICI	0-19 DL0	Dial 0 (calls from telephones in SCI 0).
ICI	0-19 DFO	Fully restricted (call from fully restricted telephones).

LD 10 – Change SCI for Analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCI	0-7	SCI number.

LD 11 – Change SCI for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
SCI	0-7	SCI number.

Feature operation

No specific operating procedures are required to use this feature.

Station Specific Authorization Code

Contents

This section contains information on the following topics:

Feature description 783
Operating parameters 784
Feature interactions 784
Feature packaging 785
Feature implementation 785
Feature operation 788

Feature description

Station Specific Authorization Code (SSAU) enables the system administrator to control the level of authorization code access on a per telephone basis. SSAU applies to analog (500/2500 type) telephones and Meridian 1 proprietary telephones; it does not apply to Basic Rate Interface (BRI) telephones.

Station Specific Authorization Code provides three levels of authorization code access:

- Authcode Unrestricted (AUTU)
  An AUTU telephone has no authorization code access limitations. Any authorization code is accepted and processed normally.

- Authcode Restricted (AUTR)
  An AUTR telephone can enter up to six assigned authorization codes. The authorization code entered must match one of the preassigned codes. Any other authorization code will be rejected and the call will not be completed.
- Authcode Denied (AUTD)
  An AUTD telephone has no access to authorization codes. Any authorization code will be rejected and the call will not be completed.

Operating parameters

The same authorization code can be assigned to more than one AUTR telephone.

There is cross-checking between LDs 10 and 11, which define a station specific authorization code, and LD 88, which ensures that the user has entered a valid authorization code.

LD 88, which is used to delete an existing authorization code, does not check if the authorization code is assigned as a station specific authorization code before the deletion.

The Station Specific Authorization Code feature does not apply when the authorization code is prompted from a tandem node.

Feature interactions

Attendant Administration

Station Specific Authorization Code does not support Attendant Administration.

Authorization Code Security Enhancement

Users cannot freely enter authorization codes from telephones that have AUTR or AUTD Class of Service.

Autodial
Speed Call

The SSAU feature treats stored autodial numbers as if they were entered at the telephone.

Feature packaging

Station Specific Authorization Code (SSAU) is package 229, which requires Basic Authorization Codes (BAUT) package 25.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 88 – Create Authorization Code data block.

2 LD 88 – Create an Authorization Code Table.

3 LD 10/11 – Activate SSAU. Use LD 10 or LD 11 according to set type.

4 LD 20 – Set Security Password.

LD 88 – Create Authorization Code data block.

Prompt	Response	Description
REQ	NEW	Create.
TYPE	AUB	Authcode data block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password.
ALEN	1-14	Number of digits in authcodes.
ACDR	YES NO	Activate CDR for authcodes. There is no default.

RANR	 0-511 0-127	RAN Route number For Large Systems For Small Systems and Succession 1000 systems
CLAS	(0)-115	Class code value assigned to authcode (NAUT).
COS	aaa	Class of Service.
TGAR	(0)-31	Trunk Group Access Restrictions.
NCOS	(0)-99	Network Class of Service.
AUTO	YES NO	Automatically generate authcodes.
- SECR	0-9999	Security password (NAUT).
- NMBR	1-9999	Number of authcodes to be generated.
- CLAS	(0)-115	Class code value assigned to authcode (NAUT).

LD 88 – Create an Authorization Code Table.

Prompt	Response	Description
REQ	NEW	Create.
TYPE	AUT	Authorization Code Table.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password.
CODE	xxxx	Authcode (number of digits must equal ALEN).
CLAS	(0)-115	Class code value assigned to authcode (NAUT).

LD 10/11 – Activate SSAU. Use LD 10 or LD 11 according to set type.

Prompt	Response	Description
REQ:	NEW CHG	Add, or modify.

TYPE:	aaaa	Telephone type, where: aaaa = 500, 2500, SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
CLS	(AUTU) AUTR AUTD	Authcode unrestricted. Authcode restricted. Authcode denied.
MAUT	(NO) YES	Modify assigned authcodes for this telephone.
SPWD	xxxx	Correct security password (if one is defined).
AUTH	x nnnn	x is in the range of 1-6; nnnn is the assigned authcode (a valid authorization code defined in LD 88).
	X x	X x deletes an assigned authcode.
Note: Changing an AUTR telephone to AUTU or AUTD clears all assigned authcode information previously defined for that telephone.		

LD 20 – Set Security Password.

Prompt	**Response**	**Description**
REQ:	PRT	Print.
TYPE:	xxx	Type of data block.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD DD 4D 8D	Card density requested.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Valid Security data password to display SSAU.
Note: Once SPWD is prompted, a valid security data password as defined in the customer data block is required for displaying Authorization (AUTH) information for sets with Class of Service Authorization Code. Sets with Class of Service Authcode Unrestricted (AUTU) and Authcode Denied (AUTD) do not have AUTH information for display. Entering of a carriage return at the SPWD prompt will result in the AUTH information being skipped during printing.		

In LD 20, Security Password (SPWD) will not be prompted if any of the following conditions exists:

- the Station Specific Authcode Package 220 is not equipped,
- the response to the TN prompt is more than one specific TN,
- the response to the TN prompt is a unique TN, but the customer of this TN does not have a security data password defined,
- the response to the CUST prompt is not a specific customer, or
- the response to the CUST prompt is a specific customer number but the customer does not have a security password defined.

Feature operation

After an authorization code is entered, the Station Specific Authorization Code feature determines if the set is allowed to use the entered code. If the authorization code is not allowed on that set, the existing invalid authorization code treatment occurs. Otherwise, normal authorization code processing occurs.

Station-to-Station Calling

Contents

This section contains information on the following topics:

Feature description 789
Operating parameters 789
Feature interactions 789
Feature packaging 790
Feature implementation 790
Feature operation 790

Feature description

Station-to-Station Calling allows direct dialing between station users in the same customer group without the assistance of the attendant.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Manual Line Service

If a single line telephone has been assigned a Manual Line Class of Service, the telephone automatically rings the attendant when it goes off-hook.

Private Lines

You must go over the public network to reach a Private Line. The software PRDN is not meant to be dialed directly.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Stored Number Redial

Contents

This section contains information on the following topics:

Feature description 791
Operating parameters 792
Feature interactions 792
Feature packaging 794
Feature implementation 794
Feature operation 795

Feature description

Stored Number Redial (SNR) allows telephones and Attendant Consoles to store one previously dialed number of 4 to 31 digits for automatic redialing.

Depending on the type of telephone, the number can be stored before a call is placed, during Ringback, while the number is busy, or during an active call. On Attendant Consoles, the number can be stored only before a call is placed. Stored Number Redial (SNR) is not supported on M2317 telephones, M3000 Touchphones, or analog (500/2500 type) telephones serving as Private Lines.

Operating parameters

When a number is stored, it overwrites any previously stored number.

Storage is limited to one number per analog (500/2500 type) telephone and one number per SNR key. When a call is established through a Tandem TIE Trunk Network (TTTN), the user is required to pause for dial tone. When you store a number using SNR, automatic redialing may fail because required delays are not added. It is possible to include delays in the outpulsing by dialing the asterisk (*) in the original digit string where dial tone is expected. Each asterisk (*) signifies a three-second delay in outpulsing.

The three-second delay is not available from a 500-type telephone.

During the stored Number Redial (SNR) programming mode, if the user attempts to store more digits than the maximum number defined for the telephone or console, SNR programming is canceled and overflow tone is returned. During an active call on a Meridian 1 proprietary telephone, if a user attempts to store more digits than the specified limit, the SNR operation fails, the previously stored number remains unchanged, and a failure indication is not given. The SNR indicator remains off.

For analog (500/2500 type) telephones, in order to store a number dialed to a busy DN, the maximum length of the stored number must be at least five digits (see prompt FTR RDL xx in LD 10).

Feature interactions

Authorization Code Security Enhancement
Charge Account
Forced Charge Account

The Authorization, Charge Account, and Forced Charge Account codes are not stored. To store a code, dial the code prior to using Stored Number Redial to dial the call.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using the Stored Number Redial (RDL) key.

Calling Party Privacy

During Stored Number Redial (SNR) programming, a user can store the Calling Party Privacy (CPP) code followed by the normal dialing sequence in the SNR data space. Outgoing calls originated by the SNR feature will send the Privacy Indicator to the far end. The CPP code will be counted against the maximum number of digits (currently 31) allowed by the SNR feature.

During an active call on a Meridian 1 proprietary telephone, the Stored Number Redial feature will store the CPP code in the SNR data space if the CPP code was included in the number dialed by the originator. The outgoing redialed calls will send the Privacy Indicator to the far end.

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Stored Number Redial.

End-to-End Signaling

End-to-End Signaling (EES) activates after a call to a trunk is established by expiration of the end-of-dial timer. Further digits dialed are not stored by the SNR feature once it is in EES mode.

Group Hunt

A Pilot DN will be stored as a Stored Number Redial (SNR) number when it is dialed directly.

Intercept Computer Dial from Directory - Post-dial Operation

An attendant can dial an extension from the Intercept Computer, and then press the Stored Number Redial key to store the called number (following the rules of the Stored Number Redial feature).

Multi-Party Operations

For analog (500/2500 type) telephones, the Last Number Redial/Stored Number Redial feature can be used when normal or special dial tone is received. The last number redialed that can be stored is the first call of a consultation connection, and can be stored only after the connection is completely released.

Feature packaging

Stored Number Redial (SNR) package 64 has no feature package dependencies.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Add or change SNR for Analog (500/2500 type) telephones.

2 LD 11 – Add or change SNR for Meridian 1 proprietary telephones.

3 LD 12 – Add or change SNR for Attendant Consoles.

LD 10 – Add or change SNR for Analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(XFD) XFA	Call Transfer (denied) allowed.
FTR	RDL xx	Activate SNR, where: xx = the maximum number of digits that can be stored (that is, 4, 8, 12, (16), 20, 24, 28, 31).

LD 11 – Add or change SNR for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, or 2616.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx RDL yy	Add an SNR key, where xx = key number, and yy = the maximum number of digits that can be stored (that is, 4, 8, 12, (16), 20, 24, 28, 31).

LD 12 – Add or change SNR for Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx RDL	Add an SNR key.

Feature operation

Attendant Consoles, Meridian 1 proprietary telephones

To store a number prior to dialing (for Attendant Consoles, and Meridian 1 proprietary telephones):

- Without lifting the handset, press **Stored No.**
- Dial the number.
- Press **Stored No.** again. The number is stored, replacing any previous one.

To store a number during Ringback, while the number is busy, or during an active call (for Meridian 1 proprietary telephones only):

- Press Stored No.

To call a stored number:

- Press **DN** (Meridian 1 proprietary telephones) or the Loop key (consoles).
- Press **Stored No**. The number is dialed.

Analog (500/2500 type) telephones

To store a number prior to dialing:

- Lift the handset.
- Dial SPRE 78, or the Flexible Feature Code (FFC) assigned for SNR.
- Dial the number to be stored.
- Hang up. The number is stored, replacing any previous one.

To store a number before a call is placed, during Ringback, while the number is busy, or during an active call:

- Flash the switchhook or press **LINK**.
- Dial SPRE 78, or the FFC assigned for SNR.

To call a stored number:

- Lift the handset.
- Dial SPRE 79, or the FFC assigned for SNR. The number is dialed.

Supervised Analog Lines

Contents

This section contains information on the following topics:

Feature description 797
Operating parameters 798
Feature interactions 799
Feature packaging 801
Feature implementation 801
Feature operation 802

Feature description

The Supervised Analog Lines feature provides two types of call supervision signaling capabilities: battery reversal answer/disconnect supervision; and hook flash disconnect supervision. These forms of supervision are provided to terminal devices connected to analog ports in the system.

Battery Reversal Supervision

Battery reversal answer and disconnect supervision signaling is used for calls originating from the terminal device. It provides both far-end (that is, the called party) answer supervision and far-end disconnect supervision signals to the terminal device. It does not apply to incoming calls terminating at the terminal device.

In the idle state, the analog port in the system provides ground signal on the tip lead and battery on the ring lead. This polarity is maintained during dialing and ringing at the far end. When the far end answers, the battery and ground connections are reversed. The reverse battery is maintained while the call is established. When the far end disconnects, the battery and ground connections are reverted to the idle state to signal that the far end has disconnected. If the terminal device disconnects first, the system sends the Deactivate Battery Reversal Scan Signal Distribution (SSD) message to the firmware after receiving the on-hook status to revert the polarity to its idle state.

Two types of battery reversal are supported. Battery Reversal for Absolute Answer Only provides an answer supervision signal to the terminal device only when the system detects an absolute answer. Battery Reversal for Absolute and Assumed Answer provides an answer supervision signal to the terminal device even when an assumed answer is detected and the far end is not capable of indicating definite answer (for example, an outgoing call on an unsupervised loop start trunk).

Hook Flash Disconnect Supervision

Hook flash disconnect supervision is used for incoming calls terminating at the terminal device. The disconnect signal is indicated by the removal of the ground connection to the tip lead for a specific period of time, which is provided by firmware ranging from a minimum of 10 milliseconds to a maximum of 2.55 seconds. The analog port is held busy for incoming calls while hook flash is in progress.

Operating parameters

This feature applies to Intelligent Peripheral Equipment that support the Supervised Analog Line feature only.

Supervised Analog Lines require NT1R20AB off premise line cards. However, NT5D11AA or NT5D14AA line cards may also be used.

Disconnect supervision is not provided to the terminal device if the system does not receive any indication of the far end releasing.

If the system does not receive any answer indication, and answer supervision is not extended to the terminal device following an assumed answer condition, disconnect supervision cannot be extended when the far end disconnects.

If the Battery Reversal Supervision feature is configured for an analog line on an analog card that does not support battery reversal, the battery reversal SSD messages from the system software are ignored by the analog card firmware. In this case, no battery reversal signal is extended to the terminal device.

If the Hook Flash Disconnect Supervision feature is configured for an analog line on an analog card that does not support hook flash, the hook flash SSD messages from the system software are ignored by the analog line card firmware. In this case, no hook flash signal is extended to the terminal device.

If the system initializes while an outgoing call originating from an analog line is established and battery reversal is activated, unprotected data is lost. In this case, battery reversal remains activated when the call is cleared down by either party. However, the line status is reverted to normal when the next outgoing call is answered and then cleared down.

If the hook flash timer is set equal to or greater than the on-hook timer, activation of the hook flash disconnect signal also causes the card to send an on-hook message and then an off-hook message to the system. In this case, if the user remains off-hook after the far end disconnects, dial tone is received and an outgoing call can be initiated.

Feature interactions

Call Transfer

If more than one active call is extended to an analog line, the call type associated with an analog line is determined by the first active call. The call type is assumed to be incoming and hook flash supervision applies if a terminal device answers an incoming call from an idle state. If the terminal device performs a switch hook flash to put the first party on hold and initiates a consultation call, the Battery Reversal feature is not supported; no battery reversal answer signal is extended to the terminal device when the second party answers.

If the first party disconnects while the terminal device is connected to the second party, no disconnect supervision is extended to the terminal device. However, hook flash disconnect supervision is extended to the terminal device when the second party disconnects (that is, a disconnect supervision signal is sent only when the last party connected to the terminal disconnects).

If a terminal device originates an outgoing call, battery reversal answer supervision is extended when the called party answers. The polarity of the line remains reversed when the terminal device performs a switch hook flash and then initiates a consultation call to a second party. The analog line is reverted to normal polarity when the terminal device completes the transfer and drops out or when the last of either the held party or the consultation party disconnects.

Conference

If a terminal device answers an incoming call and then initiates a conference, no battery reversal answer supervision signal is extended to the terminal device when new parties of the conference answer. However, a hook flash disconnect supervision signal is extended to the terminal device when the last party in the conference disconnects.

If a terminal device initiates a conference, battery reversal answer supervision is extended to the terminal device when the first party answers. No polarity change is made when additional parties are added to the conference. The polarity is reverted to normal when the terminal device disconnects or when the last party in the conference disconnects.

Multi-Party Operations

As in the cases with Call Transfer and Conference, the call type of the first active call determines whether battery reversal or hook flash supervision applies. Also, supervision signaling is not supported for the second call. A disconnect supervision signal is extended only when the last party disconnects.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 10 – Enable battery reversal supervision.

2 LD 10 – Enable hook flash disconnect supervision.

LD 10 – Enable battery reversal supervision.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
FTR	OSP (1)	Outgoing call supervision. Answer and disconnect supervision for outgoing calls with absolute and assumed answer indication. If the numeric parameter is not entered and the saved value is null, it is defaulted to 1. Otherwise it remains unchanged.
	OSP 2	Answer and disconnect supervision for outgoing calls with absolute answer supervision only.
	XOSP	Enter XOSP to disable battery reversal answer and disconnect supervision.

LD 10 – Enable hook flash disconnect supervision.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
...		
FTR	ISP 1...(75)...255	Enable hook flash disconnect supervision with flash timer in 10 millisecond units. If the numeric parameter is not entered and the saved value is null, it is defaulted to 75. Otherwise it remains unchanged.
	XISP	Enter XISP to disable hook flash disconnect supervision.

Note: Respond to the FTR prompt in LD 10 with OSP 1, and then with ISP 1...(75)...255 to enable both battery reversal supervision and hook flash disconnect supervision.

Feature operation

No specific operating procedures are required to use this feature.

Telelink Mobility Switch 1

Contents

This section contains information on the following topics:

Feature description 803
Operating parameters 805
Feature interactions 808
Feature packaging 811
Feature implementation 812
Feature operation 816

Feature description

The Telelink Mobility Switch 1 feature allows a system in conjunction with the Mobility Control Point (MCP) application to deliver a call from the Public Switched Telephone Network (PSTN) to a portable telephone subscriber. The portable telephone subscriber does not need to have a telephone physically resident in the system. A unique personal Directory Number not related to a physical termination is assigned for each subscriber of a portable. This unique personal directory number is defined as a Dialed Number Identification Services (DNIS) number within the system switch. Digit conversion is used to translate the incoming DNIS number to a Controlled DN (CDN). The CDN contains the Application Module Link (AML) number that connects the system to the MCP application.

When a call comes to the system (acting as the mobility switch) from the PSTN via a DNIS Incoming Digit Conversion (IDC) trunk, the call is terminated to a CDN by the system software through IDC operation. The last few dialed digits are saved as a DNIS (subscriber identity) number.

A CDN can be operated in controlled or default mode. If in controlled mode, call treatment is controlled by the MCP application. If in default mode, call treatment is handled by the system software and default treatment is given to the call.

When a Personal Communications Service (PCS) call terminates to a CDN which is in the controlled mode, the system will notify the MCP application by providing the call's incoming route and DNIS (subscriber identify) number. This enables the MCP application to ascertain which subscriber the caller desires to reach.

The MCP application has a table containing the last zone in which each subscriber is registered, so that the MCP application can send a message to the correct zone to find out the idle/busy status of the portable telephone subscriber.

If a subscriber is busy or unable to answer the call, the MCP application will request the system (acting as a mobility switch) to either return busy or overflow tone to the caller. If the called party has subscribed to a voice messaging service, the MCP can request the system (acting as a mobility switch) to allow the caller to leave a voice message. Meridian Mail can provide a busy tone or a no answer greeting to the caller (for example, party X is busy, would you like to leave a message?)

If the called party is idle, the MCP application will request the system (acting as a mobility switch) to optimally give ringback, provide a Recorded Announcement (RAN) or give silence to the caller, while the MCP application requests the system to make an outgoing call that will be used to alert the called party that an incoming call is waiting. This outgoing call is initiated from a phantom TN. The phantom TN does not need a physical line connection or set in the system. The phantom TN needs to be assigned as an associated set so that the MCP application will get status messages regarding the state of the phantom set. The public number of this outgoing call will be provided by a Zone Controller (ZC). The ZC reserves this incoming line which is connected to the public number.

When the phantom call is received on the reserved line, the ZC alerts the called party's portable. If MCP has requested silence for the call, then at this time the MCP will request ringback treatment for the call. Once the phantom call is answered by the subscriber, the ZC notifies the MCP application. Subsequently, the MCP application requests the system to merge the two related calls (an incoming call in the CDN queue and an outgoing call to the called party), so that the caller of the incoming call and the called party can speak to each other.

When an incoming call terminates to a CDN that is in default mode, the system (acting as a mobility switch) allows the caller to leave a voice message for the called party or give overflow tone to the caller when the call ceiling is exceeded. The CDN will be in default mode under abnormal conditions such as the AML, MCP or Application Programmable Interface (API) going down.

The Mobility Switch will also provide centralized voice prompts in lieu of zones if an exception condition is encountered when a portable is attempting to make an outgoing call.

Operating parameters

Small Systems are not supported due to the phantom TN loop capacity.

There will be a 3 DB loss on a DTI trunk when a Digital Trunk Interface (DTI) trunk is involved in a merge call, and a 0 DB loss on a Primary Rate Interface (PRI) trunk when a PRI trunk is involved in a merge call. It is therefore recommended that a PRI trunk should be used on the Mobility Switch instead of a DTI trunk.

Calls to subscribers without physical sets on the system must be originated from DNIS routes.

The MCP application should request Force Overflow to an incoming call when the DNIS information is not present in an AML-ICC message.

This feature is supported for North American markets only.

ACD-C or ACD-D reports are not a requirement of this feature. Operational measurements of PCS calls are supported by the MCP application.

External calls coming to a CDN with Value Added Server Identification (VASID) connected to the MCP application from a DID (Digital or ISDN) trunk will only be supported by this feature. For this reason disconnect supervision will be guaranteed to be returned to the system (acting as a mobility switch) when the call is disconnected.

This feature only supports TIE, CO ground start (analog, digital or ISDN) trunks as the outgoing trunk of a phantom call with ZC as the destination. For this reason, disconnect supervision must be obtained from the far end when the call is disconnected by the far end.

Combination of CDNs and ACD-DNs (Interactive Voice Response-DN) cannot exceed 240 per each customer on the system (acting as a mobility switch).

The maximum number of routes cannot exceed 512.

The maximum number of IDC/New Flexible Code Restriction (NFCR) Translation Tables per customer cannot exceed 255.

Due to the Federal Communications Commission (FCC) ruling, answer supervision is required to be returned if a “Give Silence” or “Give Music” is provided as a first call treatment to a PCS (incoming DID) call as per current operation.

If the system (acting as a mobility switch) initializes, all calls waiting in the CDN queues will be lost. The AML-INIT message will be sent to applications when an initialization occurs. When the MCP application receives an INIT message from the system (acting as a mobility switch), it erases information on existing calls.

A maximum of five Device Groups (DGRPs) will be supported per customer. An Associated Set (AST) Meridian 1 proprietary telephone with Idle Terminal for Third Party Application (ITNA) enabled can only be grouped to one DGRP.

The originator of an outgoing phantom call must be a phantom TN which is an AST Meridian 1 proprietary telephone with ITNA enabled.

An attendant set and a Basic Rate Interface (BRI) set will not be allowed to merge a call to another set or trunk.

When two trunks are joined, at least one trunk must have disconnect supervision.

The Application Module (AM Base) that interfaces with the MCP application cannot control more than one application (that is, MCP and Customer Controlled Routing applications are not supported).

A call that is initiated from the phantom set must be in established state, before it can be merged with the caller. If answer supervision is defined for the outgoing trunk, the call from the phantom set will be put in established state when the answer supervision answer is returned to the system (acting as a mobility switch). If answer supervision is not defined for the outgoing trunk, the call from the phantom set will be put in established state when the End-of-dialing timer has expired (128-32,640 msecs. after the last digit has been sent out).

If answer supervision is not defined for the outgoing trunk to which the phantom trunk is connected, it is possible that a random call may beat the phantom call to the reserved line and the caller will be given a busy tone.

If answer supervision is defined for the outgoing trunk, it is possible that the PSTN might not return the answer supervision signal to the system. If answer supervision is not returned, the system will not allow the phantom set call to be merged to the caller.

If answer supervision is defined for the outgoing trunk, it is possible that the answer supervision signal could be significantly delayed across the PSTN if the signal goes through many tandem Central Offices. This causes a subsequent delay between the time the subscriber answers the portable and the time when the incoming call is connected.

No Message Waiting Indication will be sent to the MCP application when a caller has left a voice message.

The MCP application will not know if there is an invalid mailbox (Treatment DN) used to connect to Meridian Mail.

The MCP application must return a dialable number to the system (acting as a mobility switch) to launch the outgoing call to the Zone Controller. The dialable number includes the ESN access code if necessary.

If 1+ dialing is required at the first Central Office that the phantom call goes to from the system, it should either be provided by the MCP application or inserted via digit manipulation on the mobility switch.

Enhanced Serial Data Interface (ESDI) (QPC 513 vintage G or later) or Multi-purpose Serial Data Link (MSDL) (NT6D80AA) is required to connect the system to the AM or a host.

Feature interactions

The following features interact with the Telelink Mobility Switch 1 feature:

- Report Control
- Print CDN Parameters and Options Command
- CNTL Command (determines whether CDN is in controlled mode)
- DFDN Command (sets default of ACD-DN)
- CEIL Command (controls ceiling of the CDN)
- Supervisor Control of Queue Size
- Overflow by Count
- Attendant Extension
- Attendant Recall
- Network ACD (NACD)
- Timed Overflow and Enhanced Overflow
- Display Waiting Calls (DWC key)
- Night Service
- Transition Mode via the Night Service key
- Night Mode via the Night Service key
- Incoming Digit Conversion
- Night Key Digit Manipulation
- Call Forward No Answer
- Call Forward No Answer (Second Level)
- Call Forward All Calls
- Internal Call Forward

- Feature Invocation Messages
- Hunting
- Call Forward Busy
- Remote Call Forward
- Attendant and Network Wide Remote Call Forward
- Network Call Redirection
- Call Forward Override
- Trunk Optimization
- Call Transfer
- Call Transfer – By Interactive Voice Response Unit
- Conference
- Conference – By Interactive Voice Response Unit
- No Hold Conference
- Calling Line Identification
- Basic Rate Interface
- Incremental Software Management
- PBS Set Line Disconnect
- Application Module Link Enhancements
- NCOS Restrictions
- Time-of-day Routing
- Expensive Route Warning Tone
- Off-hook Queuing
- Call Back Queuing
- Remote Virtual Queuing
- Authcode Last
- Equal Access
- 1+ Dialing

- Interchangeable Numbering Plan Area
- Inter Digit Pretranslation
- Free Call Area Screening
- New Flexible Code Restriction
- Call Forward on DumpSysload
- Flexible Numbering Plan
- Multi-party Operation
- Priority Override
- Group Hunt
- Virtual Network Services
- Originator Routing Control, and
- Enhanced Night Service.

The following are a list of features that interact with the Merge Call aspect of this feature:

- Tenant-to-tenant Access
- Class of Service Restrictions
- Network Class of Service (NCOS) Restrictions
- Trunk Group Access Restrictions
- Schedule Access Restrictions
- Trunk Barring
- Feature Group D
- Attendant Barge-in
- Attendant Break-in
- Attendant Busy Verify
- Transfer
- Conference
- No Hold Conference

- Call Waiting
- Internal Call Waiting
- Group Call
- Voice Call
- Call Park
- Station Camp-on
- Dial Intercom
- ACD-DN key
- ACD Emergency/Answer Emergency keys
- ACD Call Agent/Answer Supervisor keys
- ACD Summon Supervisor/Answer Agent keys
- Single Call Arrangement DN keys
- Multiple Call Arrangement DN keys
- HOT Line
- Private Line
- Integrated Service Access Routes
- Integrated Signaling Link
- Application Module Link Unsolicited Status Message, and
- Application Module Link Call Abandoned Message, and Digit Display.

Feature packaging

This feature is included in base System Software.

- Automatic Call Distribution, Package B (ACDB) package 41
- Network Alternate Route Selection (NARS) package 58
- Command Status Link (CSL) package 77
- Dialed Number Identification Services (DNIS) package 98
- Incoming DID Digit Conversion (IDC) package 113

- Application Module Link (AML) package 153
- Meridian Link Module (MLM) package 209
- Enhanced ACD Routing (EAR) package 214
- Enhanced Call Treatment (EACT) package 215
- Hold in Queue for Interactive Voice Response (IVR) package 218
- Call Identification (CID) package 247
- Phantom Terminal Number Operation (PHTN) package 254

The following packages are not required, but provide additional functionality:

- Call Detail Recording (CDR) package 4
- Office Data Administration System (ODAS) package 20
- ACD Load Management (LMAN) package 43
- Multi-user Login (MULI) package 242

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Two new prompts have been introduced to this overlay: ITNA and DGRP.

2 LD 17 – Configure a phantom loop.

3 LD 97 – Create a phantom superloop.

4 LD 23 – Administer a service change to a CDN data block.

LD 11 – Two new prompts have been introduced to this overlay: ITNA and DGRP.

Prompt	Response	Description
REQ:	NEW CHG MOV OUT END CHG	New, change, move, out, end, or change.
TYPE:	SL1 2006 2008 2009 2016 2018 2112	Type of Meridian 1 proprietary telephone.
CUST	xx	Customer number, as defined in LD 15
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems If l is a phantom loop and the CSL package is not equipped, an error message will be returned.
TOTN	 l s c u c u	Destination Terminal Number; prompted when REQ = MOV. For Large Systems For Small Systems and Succession 1000 systems
CFTN	 l s c u c u	Copy From Terminal Number; prompted when REQ = CPY. For Large Systems For Small Systems and Succession 1000 systems
SFMT	AUTO DN TN TNDN	DNs and TNs are assigned automatically. User enters the DN for each new telephone. User enters the TN for each new telephone. User enters DN and TN for each new telephone. Prompted when REQ = CPY.
CDEN	YES	Single, double or quad density (not prompted for superloop).
DES		ODAS designator.
...		

CLS	 (NDD) (DNDD) ...	Class of service options. No digit displayed. Dialed name display denied. Block SPV and AGN if this TN is on a phantom loop.
AST	xx yy	Associate telephone assignment for Meridian Link application.
IAPG	(0)-15	Meridian Link Unsolicited Status Message (USM) group.
ITNA	(NO) YES	Idle TN for the third party application.
DGRP	(1)-5	Device group
...		
KEY	xx SCR yyyy	Single call ringing DN key.

LD 17 – Configure a phantom loop.

Prompt	**Response**	**Description**
REQ	CHG END	Change, or end.
TYPE	CFN CEQU	Configuration Record. Gate opener.
CEQU	YES	Change Common Equipment parameters. This will be prompted if TYPE = CFN.
...		
- TERM	0-159 0-159... [X] 0-159 [C] 0-159...	Single density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMO	0-159 0-159... [X] 0-159	Single density remote terminal loops. Precede loop number with X to remove.
- TERD	0-159 0-159... [X] 0-159 [C] 0-159...	Double density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.

- REMD	0-159 0-159... [X] 0-159	Double density terminal loops.
- TERQ	0-159 0-159... [X] 0-159 [C] 0-159...	Quad density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMQ	0-159 0-159... [X] 0-159	Quad density remote terminal loops. Precede loop number with X to remove.

LD 97 – Create a phantom superloop.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop data.
SUPL	0-156 [X] 0-156 [C] 0-156	Superloop number in multiples of four. Precede loop number with X to remove a superloop. Precede the loop number with C to create a phantom superloop.

LD 23 – Administer a service change to a CDN data block.

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDN	Controlled DN data block.
CUST	xx	Customer number, as defined in LD 15
CDN	Directory Number	Controlled DN
FRRT	RAN route number	First RAN route number.
FRT	1-2044	First RAN timer.
SRRT	RAN route number	Second RAN route number.

SRT	1-2044	Second RAN route timer.
FROA	(NO) YES	First RAN to be given immediately.
MURT	Music route	Music route number.
DFDN	Directory Number	Local default ACD-DN or IVR DN.
CEIL	0-(2047)	Call ceiling value.
OVFL	(NO) YES	Force Overflow Tone to the call when ceiling threshold exceeded?
TDNS	(NO) YES	Is the DNIS number an original party?
RPRT	(YES), NO	Report on this CDN in ACD-C or D reports.
CNTL	(NO) YES	Is this CDN in controlled mode?
VSID	0-15	VASID for AML for application.
HSID	0-15	VASID for AML for host.
CWTH	0-(1)-2047	Call waiting LED threshold.
BYTH	(0)-2047	Busy queue threshold.
OVTH	0-(2047)	Overflow queue threshold.
STIO	1 2 3 ... 15	TTYs assigned for status displays.
TSFT	0-510	Telephone service factor threshold.
ACNT	xxxx	Default activity code.

Feature operation

No specific operating procedures are required to use this feature.

Telephones

Contents

This section contains information on the following topics:

Feature description 817
Operating parameters 826
Feature interactions 826
Feature packaging 826
Feature implementation 826
Feature operation 831

Feature description

There are several different types of telephones you can use in the system. Regular telephones are compatible with the system, as well as several special business telephones designed specifically to take advantage of the many features available.

This module provides an overview of the telephones and a description of the basic features and services. Additional information regarding related software features is found in other modules of this document.

Analog (500/2500 type) telephones

Analog (500/2500 type) telephones are regular telephones not normally associated with a business environment, but they are compatible with the system. They are configured by using LD 10. The 500-type telephones have a rotary dial. The 2500-type telephones are the basic push-button models, such as the Link and Unity, which do not have feature buttons normally found on business telephones.

Although analog (500/2500 type) telephones do not have feature keys, you can access various system features using Special Prefix (SPRE) codes. SPRE codes are also useful for Meridian 1 proprietary telephones to access features without using feature keys. Dial the SPRE code (unique to each customer within the system) and then the feature code that applies to the operation you desire.

Table 46 lists the feature codes available using SPRE.

Table 46
Feature codes used with SPRE (Part 1 of 3)

Dial SPRE +	Operation performed
1	Ring Again
2	Cancel Ring Again
3	Ringing Number, Call Pickup
4	TAFAS (Trunk Answer From Any Station)
5	Charge Account for CDR
6	Authorization Code Access
70 + ACOD + mmm (Trunk Route Access Code and Member)	Trunk Verification From Station
71 + DN	Call Park, To Park
72 + DN	Call Park, To Retrieve
73	System Speed Call, To Use
74	Call Forward activate or cancel (500-type telephones)
75 + Entry Access Code + DN (500-type telephones)	Speed Call, Individual To Program Entry

Table 46
Feature codes used with SPRE (Part 2 of 3)

Dial SPRE +	Operation performed
76 + Entry Access Code (500-type telephones)	Speed Call, Individual To Use Entry
77	Permanent Hold (500-type telephones)
78	Stored Number Redial, To Store
79	Stored Number Redial, To Redial
81	Automatic Set Relocation
83	Malicious Call Trace
84	Integrated Messaging System
86 + x (status)	Room Status
86 + 1	Cleaning Request
86 + 2	Cleaning In Progress
86 + 3	Room Cleaned
86 + 4	Passed Inspection
86 + 5	Failed Inspection
86 + 6	Cleaning Skipped
86 + 7	Not For Sale

Table 46
Feature codes used with SPRE (Part 3 of 3)

Dial SPRE +	Operation performed
87	Disconnect Trunk, Conference 6 (analog (500/2500 type) telephones)
89	Last Number Redial
91	Access to maintenance programs by Maintenance Telephone
92	Terminal Diagnostics, telephones and Attendant Consoles
93	Conference Circuit Testing
94	Ringing Number, Group Pickup
95	Ringing Number, DN Pickup
96	Centrex Switchhook Flash
97	Unassigned Automatic Call Distribution (ACD) analog (500/2500 type) telephone Log in/out
98	Unassigned ACD analog (500/2500 type) telephone Activate/deactivate Not Ready

Table 47
2500-type telephone features (no SPRE code used)

# + 1 + DN	Call Forward
# + 2 + Speed Call code + DN	Speed Call, Individual, To Program Entry
# + 2 + Speed Call code + *	Speed Call, Individual, To Erase Entry
# + 3 + Speed Call code	Speed Call, Individual, To Use Entry
# + 4	Permanent Hold

Meridian Modular Telephones

The Meridian Modular Telephones are designed to provide cost-effective integrated voice and data communication capability. These telephones communicate with the system using digital transmission over standard twisted-pair wiring. Table 48 summarizes the different models of Meridian Modular Telephones.

When a modular telephone is equipped with either a display or data option, a PROGRAM key (key 5 for M2006, key 7 for all remaining modular telephones) is automatically assigned to the upper right-hand feature key. This feature provides user control over such display features as screen format, contrast and language. It also provides user control over such parameters as transmission speed, parity and terminal mode.

Table 48
Meridian Modular Telephones

Set type	Programmable keys	Additional comments
M2006	6	Single-line only
M2008	8	Multi-line
M2616	16	Programmable Handsfree
M2016S	16	Telephone Security Group, Class II approved
M2216ACD-1	16	ACD Display module and two RJ-32 headset jacks
M2216ACD-2	16	ACD Display module; one RJ-32 and one PJ-327 headset jack

The Meridian Modular Telephones are designed to accommodate various add-on modules to increase their functionality. Table 49 lists the modules you can add on to a Meridian Modular Telephone.

Table 49
Add-on modules for Meridian Modular Telephones

	M2006	M2008	M2016S	M2616	M2216ACD-1	M2216ACD-2
Display		x	x	x	Standard	Standard
Key Expansion Module			x	x	x	x
Programmable Data Adapter	x	x	x	x	x	x
External alerter interface	x	x		x	x	x

Note: In this table, x indicates available add-ons for the telephone listed along the top row.

M2006

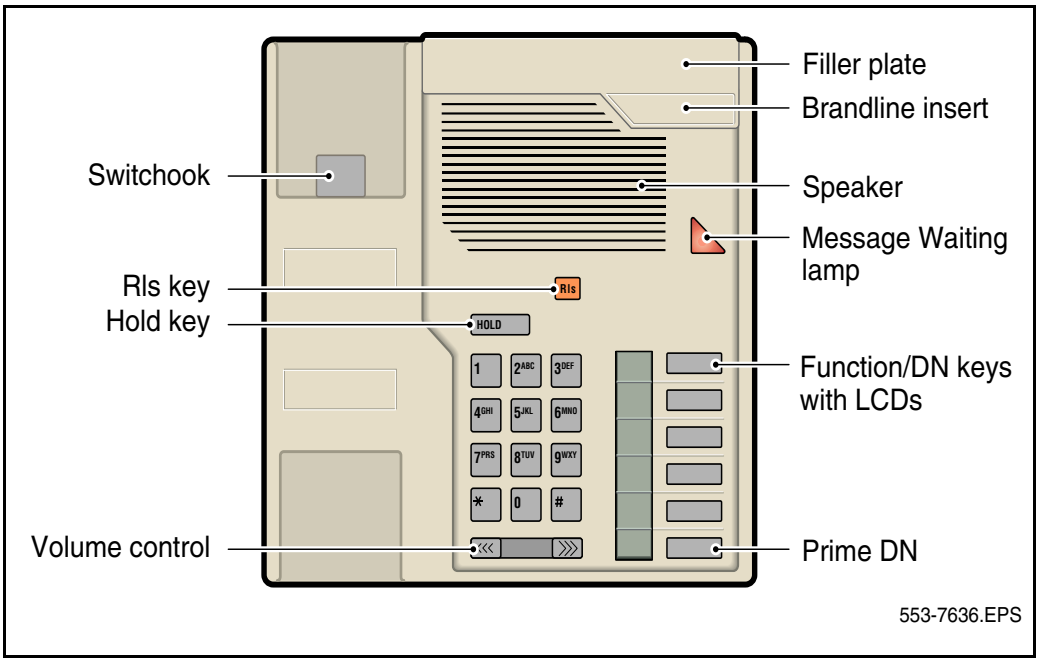
The M2006 is a digital single-line telephone that provides on-hook dialing, volume control, Release and Hold keys and a Message Waiting indicator. In addition, it provides four or five programmable feature keys (five if the data option is not in use). It also has a one-way speaker and a programmable data option.

The M2006 can have an optional external alerter interface, which connects to any standard remote alerting device.

The M2006 works off any digital line card.

Figure 19 on page 823 shows the M2006 telephone.

Figure 19
M2006 telephone



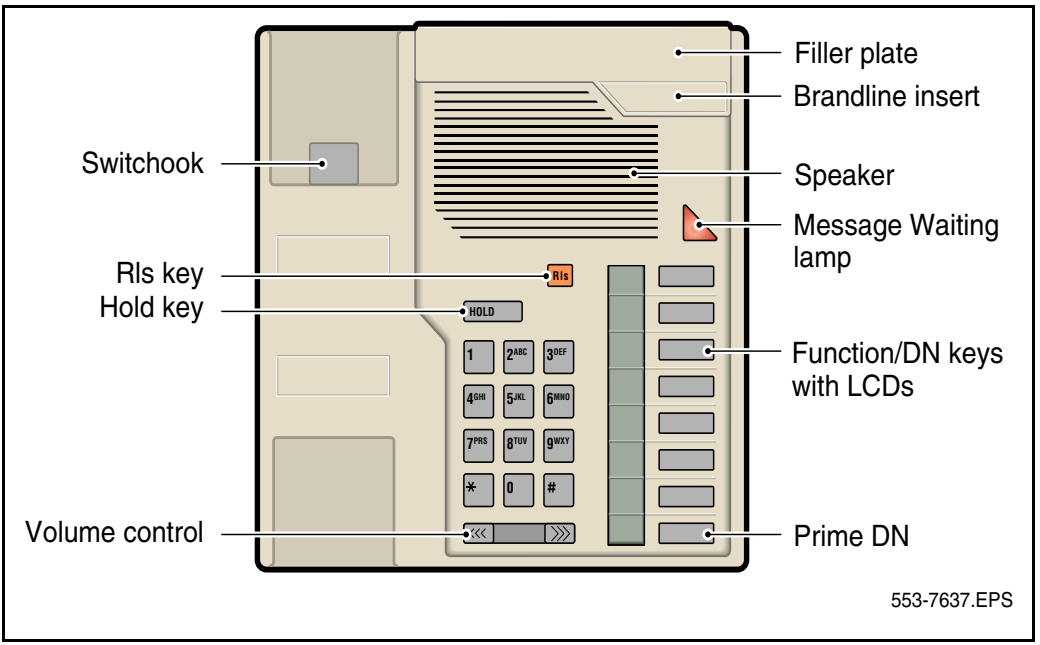
M2008

The M2008 digital telephone has eight programmable feature/line keys, on-hook dialing, volume control, Release and Hold keys and a Message Waiting indicator.

The M2008 also supports the programmable data adapter, alphanumeric display and external alerter interface options.

Figure 20 on page 824 shows the M2008 telephone.

Figure 20
M2008 telephone



M2616, M2216 (Models 1 and 2)

The M2616 telephone has 16 programmable feature/line keys, on-hook dialing, volume control, Release and Hold keys, Message Waiting indicator and Handsfree/mute features. It supports up to two add-on modules (each of 22 keys), an alphanumeric display option (two lines of 24 characters each), programmable data adapter and an external alerter interface.

The M2216 Model 1 and the M2216 Model 2 are almost identical to the M2616 with the following exceptions:

- They have no switchhook because they are designed for plug-in handset or headset operation.
- Display is standard rather than optional.

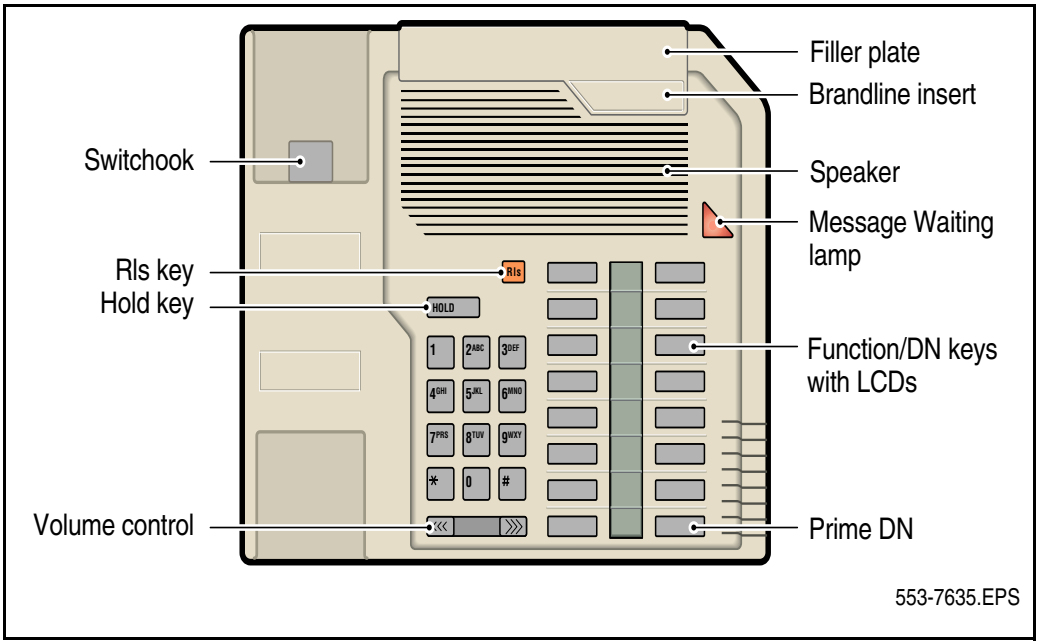
Model 1 and Model 2 refer to the types of headsets with which the M2216 operates.

Enhancements to M2216 Voice Parameters

The receive voice parameters on the M2216 can be increased up to 6 decibels. To program this capability, the ACD Set Objective Loudness Rating (AOLR) must be configured in LD 17. See "LD 17 – Change Meridian Modular Telephone transmission parameters." on page 829.

Figure 21 shows the M2616 telephone.

Figure 21
M2616 telephone



M3900 Meridian Digital Telephones

For information on M3900 Meridian Digital Telephones, refer to *Telephones and Consoles: Description* (553-3001-367).

Internet terminals

For information on the i2004 and i2050 terminals, refer to *Internet Terminals: Description* (553-3001-368).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Analog (500/2500 type) telephone and SL-1 telephone capabilities are included in base system software.

Digital Sets (DSET) package 88 has no feature package dependencies (Meridian M2000 series telephones).

M2317 telephone (DLT2) package 91 requires Digital Sets (DSET) package 88.

M3000 Touchphone (TSET) package 89 requires Digital Sets (DSET) package 88.

Meridian Modular Telephones (ARIE) package 170 requires Digital Sets (DSET) package 88, and M3000 Touchphone (TSET) package 89.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Change an existing Special Prefix Code (SPRE).

2 LD 10 – Add or change analog (500/2500 type) telephone.

3 LD 11 – Add or change Meridian 1 proprietary telephones.

4 LD 17 – Change Meridian Modular Telephone transmission parameters.

5 LD 11 – Add data TN to Meridian Modular telephones.

6 LD 11 – Add data TN to SL-1 telephones with data module.

LD 15 – Change an existing Special Prefix Code (SPRE).

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
SPRE	xxxx	Special Prefix number. The prefix must not conflict with the numbering plan.

LD 10 – Add or change analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD (DD) 4D	Card density (single, double, quad). This prompt appears only if no units on the card have been defined.
DES	a...x	Set designator (1-6 characters, alphanumeric).
CUST	xx	Customer number, as defined in LD 15
DN	xxx...x	Directory number.

TGAR	0-xx	Trunk Group Access Restriction.
CLS	aaa	Class of Service mnemonics for feature assignment.

LD 11 – Add or change Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2016, 2216, or 2616.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CDEN	SD DD 4D	Card density (single, double, quad). Not prompted for octal density. This prompt appears only if no units on the card have been defined. Card density must be 4D if TYPE is not SL-1.
DES	a...x	Designator (1-6 characters, alphanumeric).
CUST	xx	Customer number, as defined in LD 15
AOM	(0)-2	Number of key expansion modules. Prompted if TYPE = 2016, 2216, or 2616.
KLS	1-7	Number of key/lamp strips (SL-1 telephones only).
TGAR	xx	Trunk Group Access Restriction.
CLS	aaa	Class of Service mnemonics for feature availability.

KEY	xx aaa yyy...y	DN and feature key assignment (key number, feature mnemonic, directory number if applicable).

Note 1: A Message Waiting Allowed (MWA) Class of Service must be defined to enable the message waiting lamp.

Note 2: Key 7 (key 5 for M2006) is reserved for the PROGRAM key (M2008, M2016S, M2216ACD, M2616) only if display or data is equipped.

LD 17 – Change Meridian Modular Telephone transmission parameters.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATRN	Gate opener.
ATRN	(NO) YES	Change transmission parameters.
- CODE	(0)-2	CODEC coding law.
- SOLR	0-(1)-4	Sidetone Objective Loudness Rating.
- ROLR	(0)-63	Receive Objective Loudness Rating.
- AOLR	(0)-12 32-50	2216 ACD Set Receive Objective Loudness Rating.
- TOLR	(0)-63	Transmit Objective Loudness Rating.
- AGCD	(NO) YES	Automatic Gain Control disabled.

Note: Default settings are recommended. See *Transmission Parameters* (553-3001-182) before changing these parameters.

LD 11 – Add data TN to Meridian Modular telephones.

Prompt	Response	Description
REQ:	NEW	New.
TYPE:	aaaa	Telephone type, where: aaaa = 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.

TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
DES	a...x	Set designator (1-6 characters alphanumeric).
CLS	aaa	Class of Service mnemonics for feature availability.
DTYP	IOS	Inbound/outbound data station.
ADCP	(NO) YES	All digital connection prefix.
KEY	xx aaa yyy...y	DN and feature key assignment (key number, feature mnemonic, directory number if applicable). ***Note:*** Recommended key assignments for data TN are: Key 0 = DN (for data) Key 1 = DN (secondary) Key 2 = TRN (Transfer) Key 3 = ADL xxxx (Auto Dial Directory Number) Key 4 = RGA (Ring Again) Key 5 = SSC, Sc u, SSC, SSU (Speed Call, System Speed Call, controller or use – not available on M2006), and Key 6 = DSP (Display key for M2008, M2016S, M2216ACD, M2616).

LD 11 – Add data TN to SL-1 telephones with data module.

Prompt	**Response**	**Description**
REQ:	NEW	New.
TYPE:	SL1	SL-1 telephone.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CUST	xx	Customer number, as defined in LD 15
CLS	WTD	Warning Tone Denied.

KEY	xx aaa yyy...y	DN and feature key assignment (key number, feature mnemonic, directory number if applicable). ***Note:*** Recommended key assignments for data TN are: Key 0 = DN (for data) Key 1 = DN (secondary) Key 2 = TRN (Transfer) Key 3 = ADL xxxx (Auto Dial Directory Number) Key 4 = RGA (Ring Again) Key 6 = SSC, SSU (Speed Call controller or user), and Key 9 = Rls (Release).

Feature operation

Refer to the appropriate Telephone User Guide for information on telephone set operation.

Teletype Terminal Access Control in Multi-customer Environment

Contents

This section contains information on the following topics:

Feature description 833
Operating parameters 834
Feature interactions 834
Feature packaging 834
Feature implementation 835
Feature operation 835

Feature description

This is an enhancement of password usage for the Limited Access to Overlays feature. Under the previously enhanced operation, if no teletype terminal (TTY) activity had occurred for 20 minutes, the system automatically logged off. This value could not be changed. Counters were used to record the number of login attempts made on each TTY. If the threshold for the number of invalid attempts was exceeded, the system rejected any further activity at that port, for a defined period of time. No alarm mechanism was activated. Any attempt to log into the system during this period of lockout was recorded by the system.

The prompt (LOUT) in LD 17 allows the TTY administrator (PWD2 user) to define a period of time (1-20 minutes) after which the system automatically logs out if no terminal activity has occurred.

The recording of invalid attempts remains the same as before. However, if the threshold for the number of invalid entries is reached, an alarm is activated; this alarm is in the form of the “minor alarm” lamp being lit on Attendant Consoles for all customers of the system. As was the case for the previously enhanced operation, an OVL400 message is sent to all active maintenance ports and to the first TTY administrator that logs in. Other treatments also remain the same.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Intercept Computer

The Intercept Computer (ICP) feature uses maintenance LD 51 to update the system with the intercept service interface information that it stored. This overlay logs off after five minutes if no messages have been received from the Intercept Computer. This five-minute period takes precedence over the value entered in response to the LOUT prompt in LD 17. If this value is less than five minutes, the system will wait for five minutes before logging off.

Feature packaging

International Supplementary Features (SUPP) package 131; Limited Access to Overlays (LAPW) package 164.

Feature implementation

LD 17 – Configure TTY Access Control parameters.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PWD	Configuration Record. Gate opener.
- NPW1	xxx	New Password 1
- LOUT	1-(20)	Enter the time, in minutes, after which the system logs off if no terminal activity is detected.
...		
- FLTH	(0)-7	Enter the threshold for failed log-in attempts.
- LOCK	0-(60)-270	Enter, in minutes, the time the port is locked out once the FLTH has been reached.
- FLTA	(NO) YES	Enter YES to have the alarm activated once FLTH has been reached.
- AUDT	(NO) YES	Enter YES to have an audit trail activated for password usage.
- - SIZE	(50)-100	Prompted if AUDT = YES. Enter the size of the audit trail buffer.
- LLID	(NO) YES	Enter YES to activate the display of the last failed log-in attempt usage.

Feature operation

No specific operating procedures are required to use this feature.

Telset Call Timer Enhancement

Contents

This section contains information on the following topics:

Feature description 837
Operating parameters 838
Feature interactions 838
Feature packaging 838
Feature implementation 838
Feature operation 838

Feature description

The Meridian digital telephones have displayable call timers, which start after the End-of-dialing (EOD) time out expires, and not when the called party answers. With this enhancement, the call timers on these telephones do not start until a true answer is detected on all trunks with answer supervision. These include the following:

- internal stations and attendants
- ground start and loop start supervisory trunks
- Direct Inward Dialing (DID) and Direct Outward Dialing (DOD) trunks
- Digital Trunk Interface (DTI) trunks
- Primary Rate Interface (PRI) trunks, and
- TIE trunks.

On trunks without answer supervision, the call timer starts at the EOD time out.

The feature operates in standalone or Integrated Services Digital Network (ISDN) environments.

Operating parameters

There are no operating parameters associated with this feature..

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Three-Wire Analog Trunk – Commonwealth of Independent States

Contents

This section contains information on the following topics:

Feature description 839
Operating parameters 840
Feature interactions 842
Feature packaging 846
Feature implementation 846
Feature operation 863

Feature description

The Three Wire Analog Trunk – Commonwealth of Independent States (CIS) feature provides the connectivity between the system and the three-wire analog trunks (3WT) used in the CIS. Analog incoming local three-wire trunks, analog incoming toll three-wire trunks, and analog outgoing Direct Inward Dialing (DID) three-wire trunks can be connected to the system.

Cards supported in an Intelligent Peripheral Equipment (IPE) environment are referred to as X3W cards. The following XW3 cards are supported by the Three-Wire Analog Trunk – CIS feature:

- NT5K60AA for incoming local and toll trunks
- NT5K61AA for outgoing trunks

The following functions are provided by the Three-Wire Analog Trunk – CIS feature:

- Delivery of Automatic Number Identification (ANI) on request from the Public Exchange/Central Office for outgoing 3WT analog calls
- Downloading of specific transmission parameters (that is, pad data, public network toll access code, and hardware ID) for X3W cards, and
- Provision of dial tone internally by the system to the originator of the call after seizure of an outgoing X3W trunk.

The trunk state change validation timing is performed by the 3WT cards. For 3WT trunks, the originating party controls the disconnection of a call. When the originating party goes on-hook, the call is released. Note however, that when Malicious Call Trace is enabled, the Local Exchange may require a two-way release. This two-way release applies only on a telephone.

A 3WT Unproductive Timer is used to prevent a call on a X3W trunk from remaining unanswered for too long. This timer can be set to a maximum of 10 minutes.

For outgoing calls, digits are sent from the main Central Processing Unit (CPU) to the 3WT firmware. This is done by Dual-tone Multifrequency (DTMF) signaling for E3W equipment, and by IPE messaging for X3W equipment. The firmware then sends the digits as pulses and controls the actual decadic outpulsing.

Digits for incoming calls are received by the 3WT firmware as pulses. For E3W equipment, each valid pulse is reported to the main CPU by Scan and Signaling Distributor (SSD) messages. For X3W equipment, the pulses are collected by firmware and complete digits are reported to the main CPU as IPE digit messages.

Operating parameters

X3W trunk cards can only be configured on IPE shelves.

Trunk-to-trunk connections are supported, but the Automatic Number Identification (ANI) information will refer to the ANI DN of the incoming route, except with QSIG, Q931, and Digital Private Signaling System #1 (DPNSS1) routes. QSIG, and Q931 ANI information will use the Calling Line Identification (CLID) information, whereas DPNSS1 ANI will use the Originating Line Identifier (OLI) information if this information is present.

The Dynamic Loss Switching feature is not supported, because there is no connection matrix and loss alternative table available for the CIS market. However, Dynamic Loss Switching is supported in Australia, New Zealand, Italy, and China.

The Static Loss Plan Download (SLPD) feature is supported on X3W trunks.

No loss downloading/switching is done for E3W trunks.

ANI is only supported for outgoing calls.

The data in ANI is built only once at the beginning of the call. Once the trunk access code is dialed, the ANI information is downloaded to the 3WT firmware. The download of ANI occurs only once and is not changed or redownloaded for any kind of operation during a call; therefore, if the call goes through any type of modification such as a transfer or call forward for instance, the ANI information sent when requested is that of the original originator of the call.

Toll Operator Manual Ringing and Break-In are not supported on IPE analog trunks.

Data calls are supported, but with the limitations due to the 500 Hz ANI requests that can happen any time during the call and the ANI information being sent on the same voice circuit on which the data is being transmitted; therefore, the transmission of data is not guaranteed.

Multifrequency Shuttle signaling is not supported on X3W trunk cards.

The CIS A-law XCT (NTD17AE) is required.

Feature interactions

Authorization Code

An extension may, referring to the Authorization Code, seize an outgoing CIS 3WT trunk. The Authorization Code category is used to build the ANI message, meaning that a set which has a CIS restricting call category can complete a call to the public network using the Authorization Code.

Autodial

Autodial on a E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange/Central Office, which is expected to appear after the toll access code is dialed. The Public Exchange then does not accept the call due to failure to receive ANI information.

Dial Tone Detection

Dial Tone detectors are supported with the limitations of the reliability of the tone provided by the Public Exchange.

DPNSS1 Gateway

The ANI information transmitted for this incoming DPNSS1 route will include the Local Exchange Code (LEC) of the CIS outgoing route, the ANI DN, and the Category Code (CAC) of this incoming route.

The ANI DN information which is built will refer to the Originating Line Identifier (OLI) if present and the Route DN Length prompt for ANI (RDNL > 0) in LD 16. If the OLI is available, but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the OLI is available, but RDNL = 0 and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the OLI is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the Additional Digit (ADDG). If RDNL > 0, its value will be the number of digits extracted from the OLI to be used as the ANI DN. The least significant digit of the OLI will be extracted (for example, if the DN is 4201, the 1 is the least significant digit.)

If there is no OLI, the ANI DN of the DPNSS1 route is used to build the ANI message. If there is no ANI DN on the DPNSS1 route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDGs of the CIS route (ADDG is always defined).

Incoming Digit Conversion Enhancement
Incoming DID Digit Conversion

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the set, not necessarily the DID number to dial to reach the set. Therefore, if an external party uses a DN for making a call to the corresponding extension which is delivered in an ANI message, the call may fail.

Last Number Redial

Last Number Redial on an E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Multiple Appearance Directory Number

Since the ANI category is defined on a per set basis for Three Wire Analog Trunks, two stations with the same multiple Appearance DN can be assigned different ANI categories.

Q931 Gateway/BRI Gateway

The ANI information transmitted for this incoming Q931 route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

The ANI DN information which is built will refer to the Calling Line Identification (CLID) if present and the Route DN Length prompt for ANI (RDNL > 0) in LD 16. If the CLID is available but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the ADDG. If RDNL > 0, its value will be the number of digits extracted from the CLID to be used as the ANI DN. The least significant digits of the CLID will be extracted (for example, if the DN is 4201, the 1 is the least significant digit).

If there is no CLID, the ANI DN of the Q931 route is used to build the ANI message. If there is no ANI DN on the Q931 route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDG of the CIS outgoing route (ADDG is always defined).

QSIG Gateway

The ANI information transmitted for this incoming QSIG route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

The ANI DN information which is built will refer to the Calling Line Identification (CLID) if present and the Route DN Length prompt for ANI (RDNL > 0) in LD 16. If the CLID is available but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the ADDG. If RDNL > 0, its value will be the number of digits extracted from the CLID to be used as the ANI DN. The least significant digits of the CLID will be extracted (for example, if the DN is 4201, the 1 is the least significant digit).

If there is no CLID, the ANI DN of the QSIG route is used to build the ANI message. If there is no ANI DN on the QSIG route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDG digits of the CIS outgoing route (ADDG is always defined).

The ANI information transmitted for this incoming QSIG route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

R2MFC Calling Number Identification

The incoming R2MFC CNI will not be tandemed if the call is outgoing to a CIS trunk. The ANI built will be the LEC of the outgoing CIS route, the ANI DN of this R2MFC incoming route if defined (otherwise it will be the ANI DN of the outgoing CIS route, or the ADDG digit), and the CAC of this incoming R2MFC route.

The category (CAC) used to build the R2MFC Calling Number Identification (CNI) for the analog, digital and Basic Rate Interface (BRI) sets is used to build the CIS ANI. The meaning of CAC is different between the R2MFC CNI signaling and the CIS signaling (analog BRI, and digital). R2MFC CAC prompt values are in the range of 0 to 10, and the default is 0. CIS CAC prompt values are in the range of 0 to 9, and the default value is 3.

If the MFC package is equipped, but not the CIST package, the CAC prompt uses the R2MFC range and default. If the CIST package is equipped (MFC package equipped or not) the CAC prompt uses the CIS range and default.

Speed Call

Speed Call on an E3W trunk will fail for toll calls. E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Virtual Network Services

Virtual Network Services is not supported on CIS trunks.

Feature packaging

The Three-Wire Analog Trunk – CIS feature is contained in Commonwealth of Independent States Trunk Interface (CIST) package 221.

The following packages are also required to implement this feature:

- Fast Tone and Digit Switch (FTDS) package 87 (only for E3W cards)
- Flexible Tones and Cadences (FTC) package 125
- International Supplementary Features (SUPP) package 131 for DID/DOD
- Flexible Numbering Plan (FNP) package 160
- Trunk Failure Monitor (TFM) package 182, and
- Meridian 1 Intelligent Peripheral Equipment (XPE) package 203 (only for X3W cards).

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 17 – Configure the system data.

2 LD 16 – Configure an incoming X3W DID route.

3 LD 16 – Configure an outgoing X3W DID route and define the toll digit using the TDG prompt.

4 LD 18 – Configure the Special Service List.

5 LD 16 – Configure an outgoing X3W DID route and define the toll access code using the SSL prompt.

6 LD 16 – Configure an incoming E3W DID route.

7 LD 16 – Configure an outgoing E3W COT route.

8 LD 14 – Add or change trunk data for X3W outgoing DID trunk.

9 LD 14 – Add or change trunk data for E3W incoming three-wire trunk.

10 LD 14 – Add or change trunk data for E3W incoming three-wire trunk.

11 LD 14 – Add or change trunk data for E3W outgoing three-wire trunk.

12 LD 10 – Add or change analog (500/2500 type) telephones for CIS.

13 LD 11 – Add or change Meridian 1 proprietary telephones for CIS.

14 LD 12 – Add or change an Attendant Console for CIS.

15 LD 27 – Add or change Basic Rate Interface (BRI) sets for CIS.

16 LD 56 – Configure dial tone, busy tone, and tone to last party.

17 LD 88 – Configure the Authcode data block.

18 LD 97 – Configure the IPE system record for three-wire trunks.

This is an example that describes how the 3WT related features are configured. Only the prompts that are significant for the Three-Wire Analog Trunk – CIS feature are mentioned.

The following features are needed to make the feature work according to this example: B34 Codec Static Loss Plan Downloading; Partial Dial Timer; End-of-Selection Busy; Tone to-Last Party; Special Dial Tones After Dialed Numbers; Trunk Barring, and Special Service List.

LD 17 – Configure the system data.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	PARM	Gate opener.
- PCML	A	System Pulse Code Modulation companding law. A-law is to be used in the CIS market.
...		
- DTRB	70	Dual-tone Multifrequency burst and interdigit pause for the Tone and Digit Switch. Pulse/Pause Ratio 70/70. For outgoing E3W cards, the preferable digitone burst time is 70 ms.

LD 16 – Configure an incoming X3W DID route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	ICT	Incoming trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0. Validation is performed by 3WT firmware.
- TIMR	GTI 128	Incoming guard timer.
- TIMR	EOD 13952	End of dial timer, default value in milliseconds.
- TIMR	DSI 11904	Disconnect supervision timer in milliseconds.
- TIMR	DDL 0	Delay Dial Timer not needed.
...		
NEDC	ORG	Near End Disconnect Control. Originating end control.
FEDC	ORG	Far End Disconnect Control. Originating end control.
CDPC	(NO)	The system is not the controlling party on incoming calls.

...		
OPR	(NO)	This is not an outpulsing route.
PRDL	YES	Partial dial timing is equipped using EOD.
EOS	BSY	Busy signal is sent on time-out.
DNSZ	(0)-7	Number of digits expected on DID routes. 0, the default, indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Busy Tone Time. Length of Busy/overflow to be returned on DID routes in seconds.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 16 – Configure an outgoing X3W DID route and define the toll digit using the TDG prompt.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.

...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ATO 128-(4992)-65408	ANI time out timer in milliseconds. For CIS outgoing trunk routes this defines the time delay performed after the outpulsing of the toll access code.
- TIMR	OGF 0	Outgoing flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
- TIMR	GTO 2944	Outgoing guard timer.
...		
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
...		
NATL	NO	North American Toll scheme.
TDG	8	Toll Digits. List of digits after trunk access code which indicate toll calls.
...		
OPR	(NO)	This is not an outpulsing route.
...		

ACKW	(NO)	Seizure acknowledge signal is not expected.
...		
LEC	0-9999999	Local Exchange Code. A value must be entered.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 18 – Configure the Special Service List.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	SSL	Special Service List data block.
CUST	xx	Customer number, as defined in LD 15
SSL	1-15	List number for Special Service List.
SSDG	xxxx	Special Service Digit or Digits (1 to 4 digits).
...		
- TOLL	YES	The SSDG entry is a toll number.
...		
SSDG	xxxx	Special Service Digit or Digits (1 to 4 digits).
...		
- SSUC	YES	The SSDG entry is a Special Service unanswered call.
SSDG	<CR>	

LD 16 – Configure an outgoing X3W DID route and define the toll access code using the SSL prompt.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
...		
SSL	1	Special Service List number.
...		
LEC	0-9999999	Local Exchange Code.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.

RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 16 – Configure an incoming E3W DID route.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	ICT	Incoming trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0. Validation has already been done by 3WT firmware.
- TIMR	OGF 0	Outgoing flash timer should be set to 0. Validation has already been done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
...		
NEDC	ORG	Near End Disconnect Control Originating end control.

FEDC	ORG	Far End Disconnect Control Originating end control.
CDPC	(NO)	The system is not the controlling party on incoming calls.
...		
OPR	(NO)	This is not an outpulsing route.
PRDL	YES	Partial dial timing is equipped using EOD.
EOS	BSY	End of selection and busy signals enabled.
DNSZ	(0)-7	Number of digits expected on DID routes. 0, the default, indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Length of busy/overflow tone to be returned on DID routes in seconds.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 16 – Configure an outgoing E3W COT route.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	COT	Central Office Trunk data block.

...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	OGF 0	Outgoing flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
- TIMR	GTO 2944	Outgoing guard timer.
...		
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
CDPC	(NO)	The system is not the controlling party on incoming calls.
...		
NATL	NO	North American Toll scheme.
...		
LEC	0-9999999	Local Exchange Code.

ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

LD 14 – Add or change trunk data for X3W incoming DID trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
XTRK	XDID	Extended Trunk Type. IPE DID trunk card.
...		
SIGL	CIS	Trunk Signaling. Three-wire CIS trunk signaling.
CIST	(NO) YES	Prompted only for incoming routes (that is, ICOG = ICT). NO = Local trunk. YES = Toll trunk.
...		
STRI	IMM	Immediate incoming start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.

CLS	(DIP) (SHL) LOL (BARD) BARA	Dial pulse (for 3WT incoming and outgoing). Line length used for pad settings. Barring (denied) allowed.

LD 14 – Add or change trunk data for X3W outgoing DID trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
XTRK	XDID	IPE DID trunk card.
...		
SIGL	CIS	Three-wire CIS trunk signaling.
...		
STRO	IMM	Immediate outgoing start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP) (SHL) LOL (BARA) BARD	Dial pulse (for 3WT incoming and outgoing). Line length used for pad settings. Barring (allowed) denied.

LD 14 – Add or change trunk data for E3W incoming three-wire trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.

...		
SIGL	EAM	Ear & mouth.
CDEN	DD	Double density.
...		
STRI	IMM	Immediate incoming start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP)	Dial pulse.

LD 14 – Add or change trunk data for E3W outgoing three-wire trunk.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	COT	Central Office Trunk data block.
...		
SIGL	LOP	Loop start.
CDEN	DD	Double density.
...		
SUPN	YES	Answer and disconnect supervision required.
- STYP	PSP	Polarity sensitive card.
...		
SEIZ	YES	Answer and disconnect supervision required.

CLS	DTN	Digitone.

LD 10 – Add or change analog (500/2500 type) telephones for CIS.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	analog (500/2500 type) telephone data block.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 11 – Add or change Meridian 1 proprietary telephones for CIS.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 12 – Add or change an Attendant Console for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	ATT 1250 2250	Console type.

...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 27 – Add or change Basic Rate Interface (BRI) sets for CIS.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DSL	Digital Subscriber Loop data block.
...		
CLS	(DNAA) DNAD	DN of set (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

LD 56 – Configure dial tone, busy tone, and tone to last party.

Prompt	Response	Description
REQ	NEW CHG PRT	Add, change, or print.
TYPE	MCAD	Master Cadence data block.
WACD	30	Cadence number. In this example entry 30 is modified.
CDNC	60 60	On-off phases for cadence.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	FCAD	Firmware Cadence data block.
WACD	30	Cadence number. In this example entry 30 is modified.

CDNC	60 60	On-off phases for cadence. 0.3 second on, 0.3 second off.
END	REPT	Repeating cycles.
- CYCS	1	On/off cycles to be repeated.
- WTON	YES	Define tones associated with the cadence.
- - TONES	158	420 Hz and -12 dB below overload.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	FTC	Flexible Tones and Cadences data block. Used to provide special dial tone after dialed number.
...		
HCCT	YES	Hardware Controlled Cadences and Tones modification of the hardware.
...		
- BUSY		Busy tone.
- - TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	30	XCT cadence number. 0.3 seconds on, 0.3 seconds off.
...		
- TLP		Tone to last party.
-- TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	30	XCT cadence number. 0.3 seconds on, 0.3 seconds off.

- TLTP	30	Tone to last party timer in seconds.
...		
SRC	YES	Source Tones.
- SRC1		CIS continuous dial tone within the range.
- - TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	0	No cadence.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	DTAD	Special Dial Tone After Dialed Number data block.
DDGT	9	The digit 9 is to be used as an outgoing local access code.
TONE	SRC1	Tone to be provided after the dialed digit 9.

LD 88 – Configure the Authcode data block.

Prompt	**Response**	**Description**
REQ	NEW CHG	Add, or change.
TYPE	AUB	Authcode data block.
...		
CLAS	(0)-115	Classcode value assigned to Authcode (NAUT).
...		
NCOS	(0)-99	Network Class of Service Group number.
CAC	0-9	Specifies ANI category for CIS calls.

LD 97 – Configure the IPE system record for three-wire trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	LOSP	Loss Plan Tables. Configure loss parameters for downloading.
...		
TTYP	(STAT)	Install a B34 Static Loss Plan Table.
- STYP	(PRED)	A numbered predefined table is to be used.
- - TNUM	28	28 = CIS Table.
REQ	CHG	Change.
TYPE	LOSP	Loss Plan Tables. Configure loss parameters for downloading.
...		
TTYP	(STAT)	Install a B34 Static Loss Plan Table.
- STYP	CSTM	Customize a numbered predefined table.
PWD2	xxxx	Response CSTM at STYP prompt requires a PWD2 password or a LAPW password with Loss Planning Customizing Allowed (LOSA) access. This prompt appears if the appropriate password has not been given previously.
- DIDS	Rx Tx	Enter loss levels for DID short line.
- DIDL	Rx Tx	Enter loss levels for DID long line.

Feature operation

No specific operating procedures are required to use this feature.

Time and Date

Contents

This section contains information on the following topics:

Feature description 865
Operating parameters 865
Feature interactions 866
Feature packaging 867
Feature implementation 867
Feature operation 868

Feature description

The Time and Date feature provides the capability to display or modify the system time and date from the Attendant Console. If Display Time or Display Date keys are installed on the console, pressing the respective key causes the time or date to be shown on the digit display. However, these keys only allow information to be displayed, not changed.

The Change Time or Change Date keys allow the attendant to change the time or date. When a change is made, the system clock is altered to the new values. The change keys also allow display of the time or date.

Operating parameters

The Time and Date feature is available with QCW, M1250, and M2250 consoles.

If the Change Time (MTM) and Change Date (MDT) keys are provided on a console, there is no need to for the Display Time (DTM) and Display Date (DDT) keys because the MTM and MDT keys provide the display capability. DTM and DDT keys are used when the console is only allowed to view, but not change, the time and date.

When using the MTM and MDT keys, the date must be entered in the day, month, and year format; and the time must be entered in the 24-hour clock format. This is true even if the M1250 or M2250 has selected a different date and time format.

The M1250 and M2250 consoles continuously show the time and date on line one of the display. The attendant can change the format of time and date by using the Options menu.

The M1250 attendant can also change the date and time by using the Options menu. However, this only changes the time and date on the console and does not change the system clock. The MTM and MDT keys are required to change the system clock.

The date and time are downloaded to the M2250 console from the system clock and cannot be changed by the Options menu. The change time and date keys are required.

A call cannot be answered while the display/change key is activated; however, the keys can be used once the call is established.

Feature interactions

Hold

Loops used when updating time or date cannot be put on hold.

In-Band Automatic Number Identification

If the agent presses the Time and Date (TAD) key while on an In-Band Automatic Number Identification (IANI) call, the time and date remain displayed throughout the call. To display the ANI number again, place the call on hold and retrieve it. The ANI number reappears.

Network Time Synchronization

As done with the LD 12, every time the Time and Date Attendant key is used to change the system time, a request for synchronization will be made to the Master to accurately set the seconds.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD12 – Assign Time and Date keys on Attendant Consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT 1250 2250	Console type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx DDT xx DTM xx MDT xx MTM	Add a Display Date key. Add a Display Time key. Add a Display/Change Date key. Add a Display/Change Time key. ***Note:*** The range of key numbers (xx) is 0-19 on the M2250 console, and 0-9 on all other consoles.

Feature operation

To view the Time, press **Display Time (DTM)**.

To view the Date, press **Display Date (DDT)**.

To change the time, follow these steps:

1 Select an idle loop key.

2 Press **Change Time (MTM)**.

3 Enter the time using the 24-hour clock for hours and minutes (00 00).

4 Press **Change Time (MTM)**.

5 Press **Rls**.

To change the date, follow these steps:

1 Select an idle loop key.

2 Press **Change Date (MDT)**.

3 Enter the date using two digits for day, month, and year (dd mm yy).

4 Press **Change Date**.

5 Press **Rls**.

Tone to Last Party

Contents

This section contains information on the following topics:

Feature description 869
Operating parameters 870
Feature interactions 870
Feature packaging 870
Feature implementation 871
Feature operation 871

Feature description

This feature allows a Tone to Last Party (TLP) tone to be sent to analog (500/2500 type) telephones or trunks that are in the half disconnect state. The TLP is given until the system releases the trunk, or the TLP timer (0-32 seconds) times out.

During the time that the TLP tone is given to the telephone, the telephone appears busy to all incoming calls. Camp-on is denied, and attendant Break-in, busy verify, and override are temporarily denied during this time.

If a telephone is not placed on-hook and the timer times out, the telephone is set in line lockout state, and remains so until it is placed on-hook.

A trunk is in the half disconnect state if the near-end has disconnected, but the system is still holding the trunk, waiting for a message from the far-end, or for the disconnect supervision timer to time out. Barge-in is denied while the trunk is receiving the TLP tone.

The TLP is defined in each tone table. The TLP for analog (500/2500 type) telephones is defined on a customer basis, while the TLP for trunks is defined on a route basis.

Operating parameters

The TLP tone is not given to a telephone that is receiving another tone.

This feature does not apply to service trunks, such as music, paging and recorded announcement.

The TLP tone is not given to a trunk if it is being held because of the guard timer.

Feature interactions

Multi-Party Operations

The TLP tone is not given to a telephone which has Multi-Party Operations (MPO).

Operator Call Back China #1

Operator Call Back China #1 (OPCB) has precedence over TLP.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

LD 56 – Modify or change customer's tone and ringing parameters.

Prompt	Response	Description
...		
TLP	ccc ttt x xx xx xx	Tone to Last Party.
TLPT	(0)-32	Tone to Last Party Timer (seconds). No tone is given if TLPT = 0.

Feature operation

No specific operating procedures are required to use this feature.

Tones and Cadences

Contents

This section contains information on the following topics:

Feature description 873
Operating parameters 877
Feature interactions 878
Feature packaging 879
Feature implementation 879
Feature operation 879

Feature description

A tone is the frequency and level of the sound produced while the telephone is ringing, providing dial tone, or providing feature activation tones. A cadence defines the time duration for the on and off phases of a ringing or tone cycle.

A set of basic tones and cadences is available on all systems. Flexible Tones and Cadences (FTC) package 125 allows the tones to be changed.

Basic Tones and Cadences

Special dial tone

Special dial tone is supplied by the system to indicate a request for Call Transfer, Conference, and Ring Again. Special dial tone differs from regular dial tone in that it has three 128 ms interruptions at the beginning of the tone.

Overflow tone

Overflow tone can be provided on an optional basis to a station user who tries to access a trunk group when all trunks are busy, or who attempts to access features that are unavailable to their telephone. Overflow tone is best described as a fast busy signal.

Tone buzzing

Tone buzzing is used in conjunction with such features as Call Waiting and Manual Signaling (Buzz) to alert the user by a buzz tone through the telephone's loudspeaker. This applies when the telephone is off-hook or has a headset plugged in.

Flexible Tones and Cadences

The Flexible Tones and Cadences (FTC) feature, allowing the system to adapt to the tone specifications of different countries. Tones such as dial, special dial, busy, ringback, overflow, test, normal, and distinctive ringing are hardware controlled from the Tone and Digit Switch (TDS) circuit card (see Table 50). Tones such as camp-on, call waiting, intrusion, and override are software controlled, although the basic tone is still coming from the TDS card (see Table 51).

The desired cadences for the software controlled tones are defined by providing the system with the time length of the on and off phases. Software also controls ringing for analog (500/2500 type) telephones, although the voltage is supplied by the ring generator card.

The tone data is stored in tables. Every customer and route must select which tone table to use. Table 0 is filled in with default hexadecimal codes when the first customer is created and must not be changed.

All data related to the flexible tones is kept in isolated areas called Flexible Tone tables. Software Cadence tones and Master Cadence tables have an index into the MCAD table for its corresponding software cadence.

Most of the cadences are expressed in multiples of five milliseconds (ms). Therefore, in addition to the existing 128 ms timing mark, a 96 ms timing mark is introduced by a new read only memory (ROM) pack with new firmware.

Table 50
Hardware controlled tones (Part 1 of 2)

Tone	Description
Dial tone	Indicates the system can accept dialing.
Message Waiting dial tone	Indicates a message is waiting at the message center.
Call Forward dial tone	Indicates that the user has call forwarded the phone.
Call Forward Message Waiting dial tone	Indicates that the user has call forwarded the phone and a message is waiting at the message center.
Control Dial tone	Used for broker service to indicate a control digit is required after the switchhook (only for 2500-type telephones with Digitone class of service).
Busy tone	Indicates that the called DN is busy.
Ringback tone	Given to the calling party while the called party is ringing. Also given to Central Office trunks waiting for the DN to answer.
ACD RGA Ringback tone	Given to a caller to an Automatic Call Distribution (ACD) group when entering the waiting call queue and having RGA (Ring Again).
Overflow tone	Indicates that the trunk route is busy, or the DN is blocked or disabled, or that a not-allowed action has been carried out.
LDN tone	Indicates to a Centralized Attendant Service (CAS) attendant that the incoming call is a Listed DN (LDN) call from a remote site.
Camp-On tone	Provided as an initial burst when the attendant extends a call to a busy DN that is not equipped with the Call Waiting feature.
Camp-On Confirm tone	Confirms to a CAS attendant that a call to a busy DN at remote site has camped on, or that the called DN has not answered after a specified time and the calling party has come back.

Table 50
Hardware controlled tones (Part 2 of 2)

Tone	Description
Dial “0” Recall tone	Indicates to a CAS attendant that a call is a recall occurring due to attendant recall or call forward busy to an attendant from a remote site.
Hold Confirm tone	Indicates to a CAS attendant that a call placed on silent hold has timed out and is recalling.
Test tone	Provided during testing of trunk circuits.
Distinctive Ring tone	Used to differentiate between routes.
Normal Ring tone	Provided for internal calls and incoming calls if distinctive ringing or precedence ringing is not in use.

Table 51
Software controlled tones (Part 1 of 2)

Tone	Description
Agent Observe tone	Given to an agent being observed by a supervisor.
Call Waiting tone	Indicates to a busy station that another call is coming in.
Intrusion tone	Provided when the attendant initiates the Barge-In, Busy Verify, or Break-In feature.
Override tone	Provided when a user operates the Override key and enters the conversation of a busy extension.
Observe Blocking tone	Given to the supervisor who encounters blocking while attempting to observe an agent.
Off-Hook Queuing tone	Given to the call originator when the call enters the off-hook queue.
Set Relocate tone	Given after all information needed to relocate the phone is given and proven to be correct. Also given to indicate all is correct after plugging the phone back in at the relocated Terminal Number (TN).
Telset Messaging Alert tone	Indicates to caller that Telset messaging facilities have been entered.

Table 51
Software controlled tones (Part 2 of 2)

Tone	Description
Telset Messaging OK tone	Indicates to caller that the message has been received correctly and everything is fine.
Tel Status Update tone	Indicates a successful status update process.
Special Dial tone	Indicates the availability of a special function such as Conference or Transfer.
Expensive Route Warning tone	When Automatic Route Selection is in use, indicates that all inexpensive routes are busy and an expensive route must be chosen to complete the call.
ACD Call Force tone	Indicates to the ACD agent that the current call has been disconnected and a new caller is about to be given to the agent.

Operating parameters

The tones that can be produced are limited to the tones available on the particular TDS card being used.

Gradual level change is not allowed when a tone is activated.

If the Distinctive Ringing package is equipped, and a trunk route is classmarked for that feature, the cadence chosen for each call comes from the same tone table as for a normal call. The Distinctive Ringing field determines the cadences.

If a parked call was originally distinctive, and FTC is equipped, the Call Park Recall cadence takes precedence. If FTC is not equipped, the distinctive precedence ringing is given.

Because Enhanced Flexible Tones and Cadences (EFTC) is an enhancement to Flexible Tones and Cadences (FTC), the FTC package must be equipped.

A customer option determines whether the cadence will be defined by the originating or the terminating end of the call.

Feature interactions

Audible Reminder of Held Call

This feature allows for a definable cadence as a reminder of a held call. With an analog (500/2500 type) telephone, the cadence is determined by the customer's Flexible Tones and Cadence (FTC) table for the holding party. Ringing on an analog (500/2500 type) telephone is not affected by definitions for the Incoming Route option. The cadence for the reminder, and the duration between reminder rings, is always defined within the customer's tone table.

Call Forward Reminder Tone

The Call Forward Reminder Tone feature provides a way to determine whether the call forwarding feature on an analog (500/2500 type) telephone is active. For systems equipped with the FTC package, the Call Forward Reminder Tone Allowed option gives the dial tone defined by Call Forward Dial Tone to an analog (500/2500-type) set that has Call Forward active with no message waiting and the dial tone defined by Call Forward Message Waiting to an analog (500/2500-type) set that has Call Forward active and a message waiting. To get different Call Forward and Call Forward Message Waiting reminder dial tones, it is necessary to define a distinct tone and cadence for Call Forward Dial Tone and a distinct tone and cadence for Call Forward Message Waiting in LD 56, as well as to specify Call Forward Reminder Tone Allowed in LD 15.

Call Park Recall and Group Call Ring

Recall Ring and Group Call Ring are given special entries in the FTC table. New entries are added to the FTC overlay (LD 56) to define the cadence for Meridian 1 proprietary telephones, and analog (500/2500 type) telephones. The new Recall Ring entry is used to ring a telephone when recalling a Parked Call.

Conference Warning Tone Enhancement

There are no changes to the limitations to cadence numbers entry values. The same restriction still applies.

Ringing Based on Incoming Route

Enhanced Flexible Tones and Cadences (EFTC) allows the route's tone table to determine the cadence and ringing frequency for incoming calls.

10-Phase Cadence

Programming of software controlled cadences expands with EFTC from 4 intervals to 10, offering greater versatility with the cadences and cadence phases. This affects all cadences under software control.

Feature packaging

Flexible Tones and Cadences (FTC) package 125 has no feature package dependencies.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Tones, Flexible Incoming

Contents

This section contains information on the following topics:

Feature description 881
Operating parameters 882
Feature interactions 882
Feature packaging 883
Feature implementation 883
Feature operation 884

Feature description

When a telephone is off-hook, the user is alerted to a second incoming call by a buzz tone. Flexible Incoming Tones (FIT) allows the replacement of the standard buzz tone with a buzz with an on/off cadence. This feature is defined on an individual telephone basis.

When a call is presented to a telephone in any of the following situations, a tone with a special cadence alerts the user:

- Call on DN key while busy on another DN
- Call to a station that is off-hook
- Call Park recall when station is busy on another DN
- Call on Group Call key while busy on another call

- Call Waiting, and
- Call on Dial Intercom key while busy on another call.

The buzz cadence is the same as the ringing cadence that applies to a particular kind of call. For example, if a user receives a call that is a Group Call, FIT alerts users with a buzz cadence unique to group calls. If the user receives a call on the Call Waiting key, FIT provides a buzz cadence signifying call waiting.

Operating parameters

Flexible Incoming Tones applies only to Meridian 1 proprietary telephones.

Flexible Incoming Tones does not apply to the following:

- Automatic Call Distribution (ACD) call forcing
- ACD agent receiving a call on ASP key
- ACD supervisor receiving a call on AMG key
- Manual signaling
- Signal Source activated by an Attendant Console, and
- Ring Again.

Digital telephones in Handsfree mode receive the regular buzz, even if FIT is enabled.

The telephone buzzes with a cadence only if the customer and telephone options are activated. If either option is off, the telephone receives the standard buzz.

Feature interactions

Automatic Call Distribution

If an Automatic Call Distribution (ACD) agent telephone has FIT allowed and either is off-hook in the handset mode or has the headset plugged in, the agent receives a buzz cadence when a new call is presented. If FIT is not allowed, the agent telephone receives the standard buzz tone.

Dial Intercom Groups

For Dial Intercom Group (DIG) calls with the voice (V) option, if the telephone receiving the call is busy, the user hears one buzz followed by a flashing indicator. This is how DIG works with or without FIT.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 15 – Allow or deny Flexible Incoming Tones (FIT) at the customer level.
2. LD 11 – Allow or deny Flexible Incoming Tones for Meridian 1 proprietary telephones.

LD 15 – Allow or deny Flexible Incoming Tones (FIT) at the customer level.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(SBD) SBA (DBD) DBA	FIT (denied) allowed for SL-1 sets. FIT (denied) allowed for Meridian digital telephones.

LD 11 – Allow or deny Flexible Incoming Tones for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(FITD) FITA	Flexible Incoming Tone (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Total Redirection Count

Contents

This section contains information on the following topics:

Feature description 885
Operating parameters 885
Feature interactions 887
Feature packaging 887
Feature implementation 888
Feature operation 889

Feature description

This feature allows a limit to be defined on the number of redirections within a private network allowed to a call, before the call receives intercept treatment.

Both the limit on the redirection for a call and the type of Intercept treatment that the call receives are customer-defined in LD 15. This applies to on-node and off-node redirections, and to all types of redirections.

Operating parameters

The maximum value that may be given to the Total Redirection Count (TRCNT) limit is seven.

The TRCNT is kept active until the call is established or directed to the attendant.

The TRCNT takes precedence over higher count limits placed on redirected calls, while lower count limits take precedence over the TRCNT.

It is possible to define a different TRCNT limit at each node. For this reason, it is possible for a node to receive a redirected call from another node that exceeds its TRCNT limit. In this case, the TRCNT count for the call is set to the TRCNT limit defined for the node. At least one attempt is made to terminate the call before intercept treatment is given.

For off-node operation, the TRCNT count overrides the Redirection Count (RCNT) count in the Integrated Services Digital Network (ISDN) field in the SETUP message. This implies that the count transmitted to a node is either interpreted as TRCNT or Call Redirection Threshold (RCNT), depending on the configuration at the receiving node.

For off-node calls, this feature applies only to systems using Meridian Customer Defined Networking (MCDN) signaling over ISDN Signaling Link (ISL)/ISDN TIE links. Network Attendant Service is required to route a call to an attendant at another node.

Intercept to the attendant does not count as a redirection attempt.

The following ISDN call restrictions apply:

- Tandem Threshold, which is the limit placed on the number of tandem nodes allowed in a network connection
- The Public Service Telephone Network (PSTN) Threshold, which is the limit placed on the number of PSTNs allowed in a network connection
- The Call Redirection Threshold, which is a limit on the number of times that a call can be redirected off-node. If the Total Redirection Count (TRCNT) Limit is set a value greater than zero, the ISDN field in the SETUP message transports the TRCNT information rather than the Redirection Count (RCNT) information
- The Mμ/A Law Conversion Threshold, which is a limit on the number of Mμ/A Law Conversions allowed in a network connection
- Satellite Delay Threshold, which is a limit on the number of satellite delays allowed in a network connection
- Disconnect Supervision Threshold, which limits to one the number of unsupervised trunks allowed in a network connection

Feature interactions

Call Forward No Answer and Transfer

If a call has attempted Call Forward No Answer and was extended by the attendant, the call will not be intercepted when the TRCNT limit has been exceeded. The call will continue to ring the telephone until recalled to the attendant.

If Overflow (OVF), Busy (BSY), or Source (SRCx) is configured as Intercept Treatments a call attempting Call Forward No Answer, that exceeds the Total Redirection Count limit, will not be intercepted. Further redirections are prohibited and the call continues to ring the current telephone.

Group Hunt

Group Hunt takes precedence over the TRCNT feature, in that the TRCNT limit is not applied to a Group Hunt call.

Hunt
Call Forward Busy
Call Forward All Calls
Call Forward No Answer
Second-level Call Forward No Answer

Hunt, Call Forward Busy, Call Forward All Calls, Call Forward No Answer, and Second-level Call Forward No Answer redirections are limited to the value defined in the TRCNT limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

Intercept treatment

Intercept treatment is not given if a call is a Network Automatic Call Distribution (NACD) ACD call, if a call is a Central Office trunk in Night Service (specific treatment is given rather than customer-defined intercept treatment), or if the call is a data call (overflow tone is automatically given).

Feature packaging

For inter-node operation, Integrated Services Digital Network (ISDN) package 145.

For detecting trunk type across a network, Network Attendant Service (NAS) package 159.

For attendant display, Calling Party Name Display (CPND) package 95.

For the attendant to override a redirection configuration, Attendant Break-in/ Trunk Offer (BKI) package 127.

Feature implementation

LD 15 – Configure the type of intercept treatment that the redirected call receives, and the number of times that a call can be redirected before being intercepted.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	INT	Intercept treatment options
...		
- RCLE	(ATN) OVF ATN	Redirection Count Limited Exceeded as defined by TRCL. ATN is not allowed for attendant calls. NAP is not allowed for any field for RCLE.
TYPE	RDR	Call Redirection
...		
- TRCL	(0)-7	Total Redirection Count Limit. Number of times that a call can be redirected before being intercepted. Zero means that redirection is not limited by this feature, but is limited by various configurations.

Feature operation

When the total redirection count exceeds the defined limit, the call receives the customer-defined intercept treatment. This treatment includes receiving busy indication, overflow indication, or recorded announcement, receiving one of eight special tones, or being routed to the attendant. If the call is routed to the attendant, it is presented on the Incoming Call Indicator (ICI) Intercept key and the reason for redirection is given on the console display. The attendant may then use Attendant Break-In to connect to the desired station (if the desired station is established on a call).

Trunk Barring

Contents

This section contains information on the following topics:

Feature description 891
Operating parameters 892
Feature interactions 893
Feature packaging 895
Feature implementation 896
Feature operation 900

Feature description

The Trunk Barring feature provides the option of denying or allowing a direct or modified connection between customer defined routes.

Trunk Barring works in conjunction with Route Access Restriction Tables (ARTs) defined in LD 56. Trunk Barring is applied on a route basis. The four route categories that Trunk Barring recognizes, and the types of routes in each category, appear in the following table:

Table 52
Route categories and types recognized by Trunk Barring

Route Category	Route Types
Central Trunk Office (COT)	COT, FEX, WAT
Direct Inward Dialing	DID
TIE	TIE, CAA, CAM, CSA
Other trunk types	ADM, DIC, MDM, PAG, RCD

Trunk Barring applies to all methods of connecting the trunks (for example, dialing route access, call modification, or attendant extension). A route is allocated an Access Restriction Table (ART) linked by a table number (ART number) in the Route Data Block. The ART to be used for a connection is determined by the first trunk in the connection independent of whether the trunks are incoming or outgoing. The first trunk in the connection is referred to as the Originating Trunk Connection (OTC).

A default table exists so that LD 56 does not have to be used to assign an ART number to a newly created route. If the default value for each Route Category is ART number 0, no trunk barring will occur.

Operating parameters

When activated in conjunction with the Route Access Restriction Tables, Trunk Barring prohibits previously allowed connections. Previously restricted connections cannot be lifted or circumvented by Trunk Barring.

Trunk Barring does not apply to Recorded Announcement (RAN), Music (MUS), Automatic Wake-Up (AWU), or Centralized Attendant Service (CAS) trunks as it is inconsistent with their defined purposes.

Feature interactions

Access Restrictions

Trunk Barring is at the top of the hierarchy for access restrictions.

Attendant Break-In

Trunk Barring does not result in intercept treatment for Toll Operator Break-In.

Attendant-Extended Calls

When an attendant attempts to extend an Originating Trunk Connection on a barred route, overflow tone is given.

Call Forward All Calls
Call Forward Busy
Call Forward by Call Type
Call Forward External Deny
Call Forward, Internal Calls
Call Forward No Answer
Call Forward No Answer, Second Level
Call Forwarding

If an Originating Trunk Connection is forwarded to a barred route, the caller receives the intercept treatment specified in the Customer Data Block.

Call Transfer

The originator of a call transfer, unless otherwise restricted, is able to connect to a denied party on a consultation basis. Operating the Transfer key on a Meridian 1 proprietary set or going on-hook on an analog (500/2500 type) telephone does not result in a call transfer if the Originating Trunk Connection is barred. The user of a Meridian 1 proprietary set remains connected to the denied party until releasing the connection and returning to the held Originating Trunk Connection. The user of an analog (500/2500 type) telephone is re-rung by the Originating Trunk Connection when a transfer is attempted and denied.

Conference Calls

The originator of a conference call can only connect to a barred route on a consultation basis. A switchhook flash from an analog (500/2500 type) telephone results in a re-established connection with the Originating Trunk Connection. The user of a Meridian 1 proprietary set must release the barred connection to return to the Originating Trunk connection, or the conference containing the Originating Trunk connection; operating the Conference key on a Meridian 1 proprietary telephone has no effect. An attendant can return to the Originating Trunk Connection, or the conference containing the Originating Trunk Connection, by releasing the barred connection. This is done by pressing the RLS DEST key; pressing the Conference key has no effect.

Direct Trunk Access

When an Originating Trunk Connection attempts a trunk connection to a route which is restricted by its Access Restriction Table, the connection is not allowed. The intercept treatment specified in the Customer Data Block is applied.

Enhanced Night Service

Any incoming call that is routed by Enhanced Night Service to a set from which it is barred will not be connected. Overflow tone (fast busy) will be given to the incoming trunk instead.

Intercept Treatment

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

ISDN Semi Permanent for Australia

For calls using or requesting an ISPC link, Trunk Barring is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS)

If one route is barred, the system will look for the next route in the Route List Index (RLI) and if this route is not barred, the call will go through on this route. If the second route is barred, the system will continue searching the next route in the Route List Index, until an unbarred route is found.

When implementing Trunk Barring caution must be exercised not to circumvent the intended NARS/BARS restrictions.

Toll Operator Break-In

Trunk Barring results in intercept treatment for all route types that can be barred, except Toll Operator Break-In.

Trunk to Trunk Connection

Trunk Barring takes precedence over the Trunk to Trunk Connection feature.

Virtual Network Services

With respect to this feature the following cases apply:

- When the second trunk involved in the call is used by VNS, no trunk barring is applied regardless of the configuration of the first trunk. The call is always allowed to get through.

  Note: This implementation completely overrides the Trunk Barring feature.

- When the first trunk involved in the call uses VNS, and the second one is not used by VNS, trunk barring is performed according to the content of the default ART table for the TIE trunk.

Feature packaging

This feature requires Trunk Barring (TBAR) package 132.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 56 – Modify trunk barring Access Restriction Tables (ARTs).

2 LD 56 – Change or print ART number for the route.

3 LD 56 – Change or print the route category default table.

In most cases that require barring, only one ART is necessary, although multiple ARTs may be defined per route. Whenever a new route is created (in LD 16), the default ART defined for that route type is assigned to the route. This default depends on the route type being created.

The flexibility of assigning ART by route is also available. The default table which specifies which ART table is to be assigned to a route type is changeable in LD 56. Until this is done, the default ART is used.

The following is a guideline on how to set up Trunk Barring:

1 Gather all information regarding the type of route to be used in the system.

2 For each route type, list beside it the route types that are barred from connecting to it.

3 For each route type, assign a code number from 1 to 63. Look for the route types that are barred from accessing the same types and assign the same code number to them. If a route type is not barred from accessing any other route type, it is assigned code number 0.

4 When each route type is assigned a code number, go back to step 2 and replace the route types that the route is barred from accessing with their code number.

5 Using LD 56, create all necessary Access Restriction Tables (ARTs). Using the code number of the originating route type as the ART number, deny the necessary route type using the code numbers assigned in the previous step.

6 Assign each ART to a route in one of two methods:

7 Use LD 56 to create the Route Category Default Table (RCDT). As each route is created using LD 16, it is assigned the default ART according to route type.

8 Use LD 56 to assign to existing routes the desired ART.

The following is an example of how to set up trunk barring using the procedures listed above. This example is not reflective of the typical situation, but is only used to show the steps involved.

List all route types.

- COT
- TIE
- DID
- PAG
- DIC
- RAN – ignore because it cannot be barred.
- MDM

List route types to which the originator is barred access.

- COT is not barred from accessing any type.
- TIE is barred from accessing COT, PAG, DIC, and DID.
- DID is barred from accessing TIE, DIC, and MDM.
- PAG cannot be originator, but can be barred by other route types.
- DIC cannot be originator, but can be barred by other route types.
- MDM is barred from accessing COT, DID, PAG, and DIC.

Assign each originating route type a code number from 0 to 63.

- COT is assigned 0 (it is not barred access to any route type).
- TIE is assigned 1.
- DID is assigned 2.
- PAG is assigned 0 (this cannot be an originating route, but it can be barred by other route types).

- DIC is assigned 0 (this cannot be an originating route, but it can be barred by other route types).
- MDM is assigned 3.

Replace the route types the originator is barred from accessing with their code numbers.

- COT (ART 0) – not barred.
- TIE (ART 1) – is barred from accessing 0, and 2.
- DID (ART 2) – is barred from accessing 0, 1, and 3.
- PAG/DIC (ART 3) – not barred.
- MDM (ART 4) – bars 0, and 2.

Set up the Route Category Default Table (RCDT).

- COT 0.
- TIE 1.
- DID 2.
- OTH 0 – MDM will initially be assigned ART 0 like DIC and PAG, but can be changed using the RART prompt.

LD 56 – Modify trunk barring Access Restriction Tables (ARTs).

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TBAR	Add or change Access Restriction Table(s) (ARTs).
ART	(0)-63	Select ART to add or change. If ART table 0 is defined, no restrictions apply.
	<CR>	Return to REQ prompt.

DENY	yyy yyy ALL xALL Xyyy Xyyy <CR>	Enter ART numbers denied to Originating Trunk Connection (OTC). Deny all ARTs to OTC. All ART numbers allowed to OTC. Enter ART numbers allowed to OTC, or change to remove previously blocked connections. Return to REQ prompt with no table being stored.

LD 56 – Change or print ART number for the route.

Prompt	Response	Description
REQ	CHG PRT	Change or Print. ***Note:*** REQ = NEW, or OUT is disallowed for RART.
TYPE	RART	Change ART number for the route.
CUST	xx	Customer number, as defined in LD 15
ROUT	(0)-511 (0)-127	Route number. For Small Systems and Succession 1000 systems
ART	(0)-63 <CR>	ART to assign to route(s). If ART table 0 is defined, no restrictions apply. Return to REQ prompt. ART remains unchanged.

LD 56 – Change or print the route category default table.

Prompt	Response	Description
REQ	CHG PRT	Change or Print. ***Note:*** REQ = NEW, or OUT is disallowed for RCDT.
TYPE	RCDT	Change the route category default table.
COT	(0)-63	COT, FEX, and WAT routes are assigned the entered ART when the route is created in LD 16.
DID	(0)-63	DID routes are assigned the entered ART when the route is created in LD 16.

TIE	(0)-63	CAA, CAM, CSA, and TIE routes are assigned the entered ART when the route is created in LD 16.
OTH	(0)-63	ADM, DIC, MDM, PAG, and RCD routes are assigned the entered ART when the route is created in LD 16.
	<CR>	Return to the REQ prompt.

Feature operation

Barring is implemented via service change by a qualified technician. If the connection is not allowed, intercept treatment defined by the ACCD prompt in LD 15 is implemented.

Trunk Failure Monitor Enhancement

Contents

This section contains information on the following topics:

Feature description 901
Operating parameters 901
Feature interactions 902
Feature packaging 902
Feature implementation 902
Feature operation 902

Feature description

This enhancement to the Trunk Failure Monitor feature provides a visual display on attendant consoles to indicate Direct Inward Dialing (DID)/Direct Outward Dialing (DOD)/TIE trunk line-break alarm conditions, and optionally to indicate 2.0 Mbps Digital Trunk Interface or Primary Rate Interface (PRI) Out-of-service conditions. The upper-most left key lamps on the console flash to indicate these trouble conditions.

This capability is available on the following Attendant Consoles:

- M1250
- M2250

Operating parameters

Trunk Failure Monitor (TFM) package 182 must be equipped.

This enhancement is not supported for:

- Tenant Groups attendants
- 1.5 Mbps Digital Trunk Interface (DTI), and
- Automatic Trunk Maintenance.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires Trunk Failure Monitor (TFM) package 182.

Feature implementation

LD 15 – Configure the attendant trunk failure display.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options
...		
- TFDR	(NO) YES	Trunk Failure Display required. Prompted with TFM package 182. Requires QCW3, QCW, M1250, or M2250 consoles.

Feature operation

The upper-most left key lamps on the console flash to indicate trouble conditions.

If the attendant is in Position Busy, Night Service, or Loop Busy state, without a call on the console, pressing the upper-most left key causes the display to show the failed trunk unit or loop number. The lamp state changes from flashing to lit. If there is more than one failed trunk or loop, the display shows them one at a time, and the lamps remain flashing until all failed trunk units or loop numbers are displayed.

When the trouble conditions have been resolved, the lamps become dark to indicate that the trunk or loop is available for normal use.

Trunk Failure Monitor

Contents

This section contains information on the following topics:

Feature description 905
Operating parameters 906
Feature interactions 906
Feature packaging 907
Feature implementation 907
Feature operation 907

Feature description

The Trunk Failure Monitor (TFM) feature detects Line Break Alarm Signals (LBAS), which are generated because of trouble conditions on Direct Inward Dialing (DID), Direct Outward Dialing (DOD), or TIE trunks, or service degraded to Out-of-service (OOS) on 2.0 Mbps Digital Trunk Interface (DTI) or Primary Rate Interface (PRI) trunks. If a line break is detected, a trunk message is printed on the maintenance TTY, and the affected trunk is rendered BUSY to stop any further seizure of the trunk during outgoing calls.

Once the line break trouble condition has been fixed, a different Line Break Alarm Signal (LBAS) is generated. The TFM feature detects this signal, prints another trunk message on the TTY indicating that the trouble condition has been corrected, and renders the repaired trunk unit IDLE for normal use.

Operating parameters

TFM is not supported by the Attendant Administration feature.

TFM is not supported on 1.5 Mbps DTI.

TFM requires the QPC730B for DID or DOD trunks, and the QPC774 for TIE trunks.

A Centralized Attendant Service (CAS) attendant can only monitor the trunks on the switch on which the attendant is located.

This feature is supported on the M2250 Attendant Console.

Feature interactions

Extended DID/DOD Software Support - Europe

As part of the Trunk Failure Monitor feature, the BAR/UNBAR messages are received from IPE XDID trunks (LD 15 must be configured with TFDR = YES); when a BAR message indicating a problem situation is received, a TRK501 message is printed on the TTY, the uppermost key lamps light up on the Attendant Console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A TRK502 message is printed on the TTY, the lamps on the Attendant Console are darkened, and the trunk is idled. Note that Baring Allowed (BARA) CLS must be configured on the XDID trunk for the described process to occur.

Extended Flexible Central Office Trunk Software Support

As part of the Trunk Failure Monitor feature, the BAR/UNBAR messages are received from IPE XFCOT trunks. When a BAR message indicating a problem situation is received, a trunk message is printed on the TTY, the uppermost key lamps light up on the Attendant Console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A message is printed on the TTY, the lamps on the Attendant Console are darkened, and the seized trunk is idled. Note that BARA Class of Service must be configured on the trunk for the described processing to occur.

Feature packaging

Trunk failure Monitor (TFM) package 182.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Trunk to Trunk Connection

Contents

This section contains information on the following topics:

Feature description 909
Operating parameters 911
Feature interactions 912
Feature packaging 914
Feature implementation 914
Feature operation 916

Feature description

The Trunk to Trunk Connection feature introduces the following capabilities: transfer on ringing of external trunk across the network, transfer of one supervised outgoing external trunk to another, conference of external trunks and outgoing trunk to trunk charging. These capabilities are available on an analog (500/2500 type) set, proprietary set or an Attendant Console.

Transfer on Ringing of External Trunk over Network

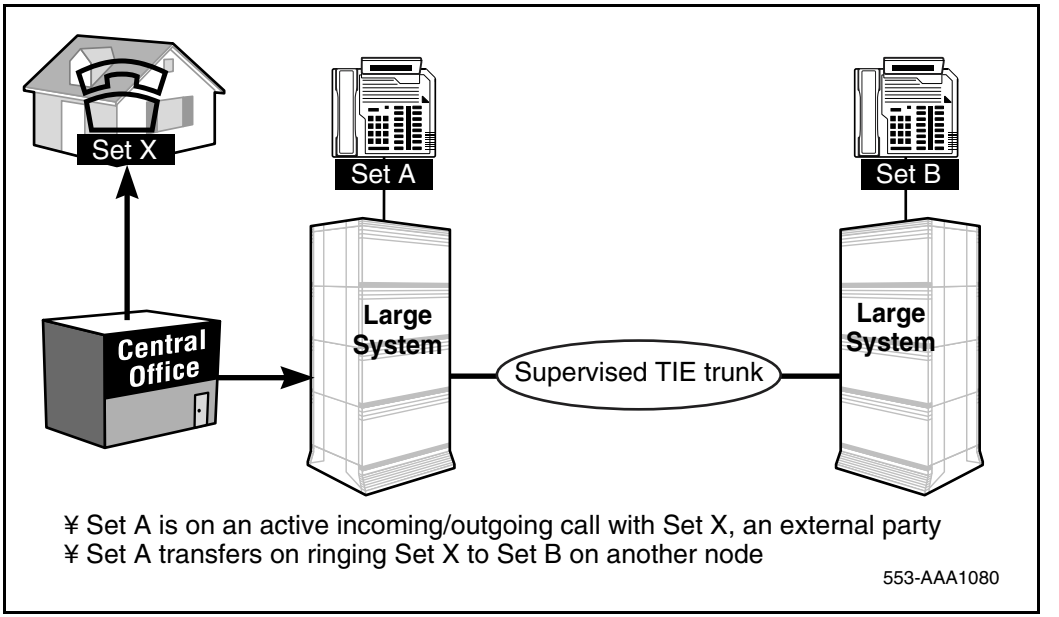
Allows the transfer on ringing of an established external trunk call over a supervised analog network TIE trunk. If the called party does not answer within a specified time, the call will slow answer recall to the attendant on the transferring node. This capability ensures that available network resources are not occupied indefinitely.

Transfer of External Trunks

Allows the transfer of one outgoing external trunk to another trunk provided both calls are answered and both trunks have answer supervision.

As illustrated in Figure 22, Set A is on an incoming/outgoing call with Set X, an external trunk. Set A initiates a call transfer of Set X to Set B. With the Trunk to Trunk Connection feature, Set A can transfer on ringing without waiting for Set B to answer. If Set B does not answer the transferred call, the external trunk will slow answer recalls to the attendant on the transferring node.

Figure 22
Transfer on ringing of external call

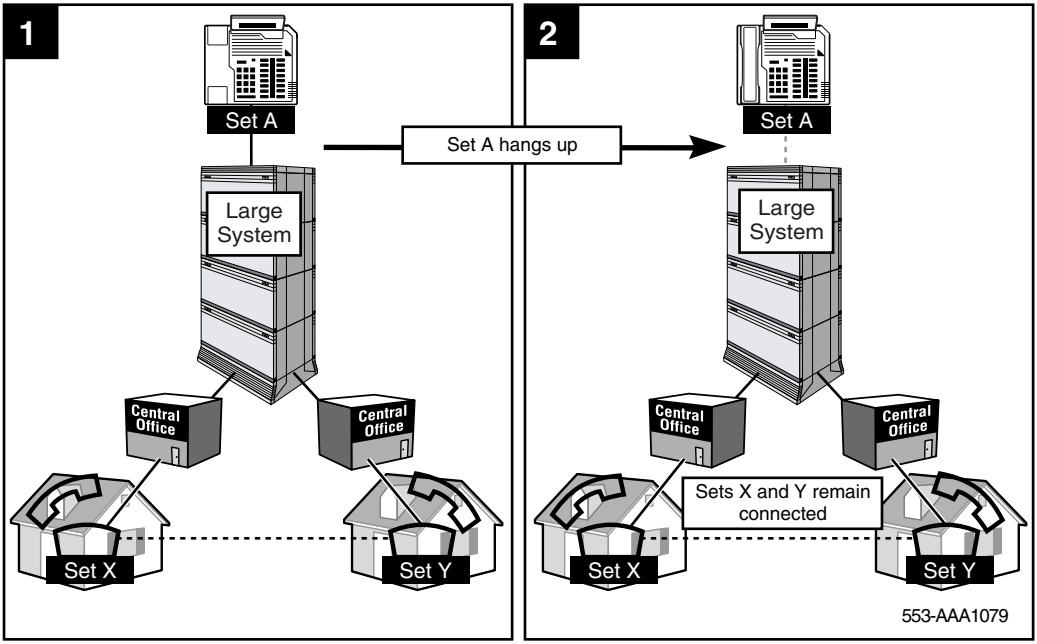


Conference of External Trunks

Allows external trunks to remain established in a conference call in circumstances when all external trunks involved in the call offer disconnect supervision.

Figure 23 illustrates the Conference of External Trunk capabilities of this feature. Set A is on an established conference call with two or more external trunks, Set X and Set Y. When Set A disconnects during the conference, Set X and Set Y continue in the established call.

Figure 23
Conference of external trunks



Outgoing Trunk to Trunk Charging

Ensures that outstanding charging information, relevant to both outgoing calls, is contained in relevant Call Detail Recording records.

Operating parameters

Slow answer recall occurs when an external trunk is transferred on ringing across an answer supervised network TIE trunk to a set that does not answer. However, the resulting recall will be to an attendant on the transferring node and not to the original set which transferred the call.

When transferring one outgoing trunk to another, it is required that the two external calls involved are both answered prior to completing the transfer. Both external trunks involved must have both answer and disconnect supervision.

When the last internal party disconnects from a conference call, involving two or more external trunks, all external trunks must have disconnect supervision for the call to remain established. If any one of the remaining external trunks does not have disconnect supervision, all external trunks will be dropped.

No change is made to existing VNS operation.

Feature interactions

Busy Tone Detection for Japan

Busy Tone Detection for Japan does not impact Trunk to Trunk Connection. However, whichever occurs first, prevails.

Call Transfer

To transfer an external trunk on ringing across a supervised analog network TIE trunk, the external trunk and internal TIE line must have both answer and disconnect supervision, and the external call must be established. To transfer one outgoing external trunk to another, both external trunks must have answer and disconnect supervision, and both external calls must be established.

Centralized Answering Position

A Small System may not have an actual attendant console. Instead, the Small System will use Centralized Answering Position (CAP). The CAP Directory Number (DN) is the customer Night DN. Since no attendant is configured, the customer is viewed to be in Night Service and any calls for the attendant are directed to the CAP. Slow Answer Recall may be presented to a CAP when no attendant console is configured for the customer.

Conference

Trunk to Trunk Connection allows external trunks to remain established in a call, provided that all external trunks involved have disconnect supervision. With respect to charging costs associated with a conference call, once the last set involved in the conference call disconnects, a search is made of all remaining trunks in the call to determine which call is established in the call for the longest period of time. This trunk is the chargeable Terminal Number (TN). This process is repeated to find the next chargeable TN.

Multi-Party Operations - Ringing No Answer

In a standalone environment, the RGNA prompt in the Customer Data Block will be used when an external trunk is transferred on ringing and the called party does not answer. In a network environment, the RTIM timer value in the Customer Data Block will be used for slow answer recall.

Message Registration

The last party releasing the call collects the total value of outstanding Periodic Pulse Metering (PPM) generated on outgoing trunks. If the last party is an internal set, the outstanding PPM is stored against the meter of the set. If the last party is an internal TIE trunk, the outstanding PPM is stored against the meter associated with the internal TIE trunk access code. If the last party is an outgoing external trunk, the outstanding PPM is stored against the meter associated with the external trunk access code.

Night Service

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

Night Service Enhancement

Recalls made while the attendant is in Night Service are routed to the Night DN, if the original call is an external call. In such a case, the destination party is disconnected, the internal network trunk is released and the original extended call is presented to the Night DN. If the original call is internal, recalls are put in the attendant call waiting queue when in Night Service.

Trunk Barring
Trunk Group Access Restriction

Trunk Barring and Trunk Group Access Restriction takes precedence over the Trunk to Trunk Connection feature.

Feature packaging

This feature is included in base System Software.

Note: DID to TIE (DTOT) for Japan package 176 must be restricted to enable this feature.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Modifications to Customer Data Block.

2 LD 15 – Modifications to Customer Data Block.

LD 15 – Modifications to Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT	Attendant console options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
RTIM	xxx yyy zzz	Enter defined value for the Slow Answer Recall timer where: xxx = 0-(30)-378 Slow Answer Recall yy y= 0-(30)-510 Camp On Recall zzz = 0-(30)-510 Call Waiting Recall

LD 15 – Modifications to Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	NET	Trunk and network options.
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
...		
ISDN	YES	Change the Integrated Services Digital Network options.
- PSTN	NO	Public Switched Telephone Network. Limit the number of PSTNs allowed in a network connection to one PSTN. NO = Put no limit on the number of PSTN connections. YES = Limit the number of PSTN connections.
...		
DITI	YES	Allow Direct Inward Dialing to TIE connections for customer.
TRNX	YES	YES = Allow transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network. NO = Prevent transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network.

EXTT	(NO) YES	NO = Prevent unconditional external Trunk to Trunk transfer. YES = Allow unconditional external Trunk to Trunk transfer. Enter YES to indicate that a caller can transfer a call made to one outgoing trunk to another outgoing trunk without restrictions based on supervision. Trunk to trunk connection features such as TGAR/TARG and Trunk Barring will still apply as appropriate. Enter NO to indicate that the transfer is allowed only if conditions regarding supervision are met. This prompt only applies to situations involving two calls originated by the same caller.

Feature operation

No specific operating procedures are required to use this feature.

Trunk Traffic Reporting Enhancement

Contents

This section contains information on the following topics:

Feature description 917
Operating parameters 918
Feature packaging 919
Feature packaging 919
Feature implementation. 920
Feature operation. 920

Feature description

The following modifications to trunk traffic reporting have been implemented to improve the accuracy of TFC002 traffic reports. The options are selected in the Configuration Data Block.

Traffic Period Option

Without enabling this option, trunk usage added its entire duration into the traffic period in which the disconnection occurred. If the duration was longer than 36 CCS (CCS = 100 call seconds), but less than 50 CCS, a TFS401 message was output. However, that duration was still accumulated and included in the traffic reports. If the duration was longer than or equal to 50 CCS, a TFS402 message was output. This duration was not accumulated, and was excluded from the traffic reports.

The Traffic Period Option enables the CCS to be reported in each traffic report interval. The peg count is still reported at disconnect time as per existing operation.

Note that when the Traffic Period Option is first enabled, the first traffic report may get some TFS403 messages.

Trunk Seizure Option

Without enabling this option, system traffic statistics began accumulating when a call was established. system software determined that the call was established when one of the following occurred: the End-of-Dialing (EOD) timer timed out after the last digit was dialed; the octothorpe (#) was dialed; or answer supervision was received. In some situations, customers could not match system traffic reports with their carrier reports, because many carriers start accumulating statistics when a trunk is seized.

The Trunk Seizure Option provides the ability to start accumulating statistics upon trunk seizure, rather than when the call is established.

Operating parameters

If the duration of a call is less than two to four seconds, the peg count is not accumulated. This functionality only applies when the trunk seizure option is enabled.

Due to the accumulation at trunk seizure, peg counts occur even if a call is unanswered.

Feature interactions

Automatic Call Distribution

A trunk call to an Automatic Call Distribution (ACD) DN will only be considered established once this call is answered. It is not considered established while the call is waiting in the ACD queue. Therefore, at the end of a traffic period, if a trunk call is in the ACD queue, the Traffic Period Option will not accumulate the duration for this call.

Note that when the duration is accumulated at disconnect or at the end of a traffic period after this call is answered, the total duration including the time the call was in the ACD queue is accumulated. This total duration may be longer than a single traffic period due to the time in the ACD queue and a TFS401, TFS402, or TFS403 message may be output.

Music Trunks

The Trunk Seizure Option is not supported on Music trunks.

Recorded Announcement Trunks

The Trunk Seizure Option is not supported on Recorded Announcement trunks.

Traffic Monitor

The Traffic Monitor feature outputs certain traffic data approximately every minute.

The trunk usage and peg count output by the Traffic Monitor feature can be enhanced by enabling the Trunk Seizure Option. The accumulated duration and peg count of a call will begin at trunk seizure time instead of at the time the call was established.

The Traffic Monitor output that starts during the same time that the regular traffic report starts is impacted if the Traffic Period Option is enabled. With this option enabled, the duration of all currently established calls is accumulated at the end of the traffic period. Therefore, this additional duration is also accumulated in the next minute's traffic monitor output. For example, the Traffic Monitor feature and the Traffic Period Option are both enabled. Regular traffic reports are output every half hour. The difference in the accumulated duration from 10:29 to 10:30 may increase dramatically due to the additional durations accumulated for currently established calls at the end of this traffic period.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 17 – Configure Traffic Reporting option.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	Gate opener.
...		
- TPO	(NO), YES	Traffic Period Option. Enter YES to enable, NO to disable, and <CR> to keep the current value.
- TSO	(NO), YES	Trunk Seizure Option. Enter YES to enable, NO to disable, and <CR> to keep the current value.

Feature operation

No specific operating procedures are required to use this feature.

Trunk Verification from a Station

Contents

This section contains information on the following topics:

Feature description 921
Operating parameters 922
Feature interactions 922
Feature packaging 923
Feature implementation 923
Feature operation 923

Feature description

Trunk Verification from a Station (TVS) provides the capability for a classmarked 2500-type telephone (that is, basic push-button set having no feature keys) to seize a particular trunk within a trunk group, receive a dial tone, and outpulse digits to complete a call to a remote maintenance site. This feature is used as part of a PC-based Network Management system to allow physical testing of each trunk in the network.

Any compatible, customer-provided PC-based circuit switched network administration and maintenance system can access the trunk to be tested by calling a remote customer-provided responder. The responder supplies the various tones needed to perform the trunk test. The PC then stores and processes the results. Once the testing is complete, the PC disconnects from the tested trunk and accesses the next trunk in the route.

To the system, the PC appears as a 2500-type telephone, which requires the capability to seize a particular trunk member within a trunk route.

Operating parameters

It is recommended that the telephone with a Trunk Verification Allowed (TVA) Class of Service also have CFW All Calls To External DN Denied (CXFD), CFW Busy Denied (FBD), and CFW No Answer Denied (FND) Classes of Service. This setup prevents any restricted telephone from accessing trunks by calling the TVA telephone and subsequently getting transferred or forwarded.

Also, it is strongly recommended that this unit not be configured with an LPA. This will prevent the unit from initiating the PBXT (test message waiting lamps) command in LD 32.

The telephone with a Trunk Verification Allowed (TVA) Class of Service should also be assigned Warning Tone Denied (WTD) Class of Service. This will prevent Attendant Busy Verification, which could impair the trunk frequency measurements that take place during a TVS call. This also prevents the trunk that this telephone has seized from being barged into by the attendant.

Trunk Verification from a Station is not applicable to B-channels on digital links.

When using the Trunk Verification feature to test network trunks, any trunk state other than an idle, such as busy, disabled or maintenance busy, an overflow tone is returned.

Feature interactions

The environment in which the TVS feature will be invoked is a machine environment. That is, the user of the 2500-type telephone with this feature will usually be a PC-based maintenance system. Therefore, minimal interaction exists with other features.

When the 2500-type telephone with a TVA Class of Service makes a TVS call, any Trunk Group Access Restrictions/Trunk Access Restriction Groups (TGAR/TARG) defined in the system are removed for this call.

When a trunk group is busied out by an Attendant Console, access to that trunk group is not allowed with the TVS feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 10 – Allow or deny Trunk Verification from a 2500 telephone.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(TVD) TVA DTN	(Deny) allow TVS. Digitone service is required for 2500 telephones.

Feature operation

To verify that a trunk is working properly (from a 2500 telephone with TVA Class of Service), follow these steps:

1 Lift the handset.

2 Dial SPRE + 70 + ACOD + mmm

where:

SPRE	is the special function access prefix
70	is the special access code for the TVS feature
ACOD	is the access code of the trunk group to be tested, and
mmm	is the number of the trunk member that is to be seized; mmm must be three digits (for example, 001).

Uninterrupted Line Connections

Contents

This section contains information on the following topics:

Feature description 925
Operating parameters 925
Feature interactions 926
Feature packaging 926
Feature implementation 926
Feature operation 928

Feature description

Uninterrupted Line Connections are connections assigned Warning Tone Denied (WTD) Class of Service. The feature prohibits the imposition of any Camp On or intrusion tones on that line.

This feature is recommended for modem or data lines.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Barge-In
Attendant Busy Verify
Override

These features cannot be applied to stations with a WTD Class of Service.

Camp-On

A call can be camped on to a station with a WTD Class of Service, but tone is not provided.

Feature packaging

This feature is included in base System Software.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1. LD 10 – Assign Warning Tone Allowed for analog (500/2500 type) telephones.
2. LD 11 – Assign Warning Tone Allowed for Meridian 1 proprietary telephones.
3. LD 14 – Assign Warning Tone Allowed for trunks.

LD 10 – Assign Warning Tone Allowed for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.

TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Warning tone (allowed) denied.

LD 11 – Assign Warning Tone Allowed for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Warning tone (allowed) denied.

LD 14 – Assign Warning Tone Allowed for trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
TN	 l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
CLS	(WTA) WTD	Warning tone (allowed) denied.

Feature operation

No specific operating procedures are required to use this feature.

United Kingdom Analogue Hardware Support

Contents

This section contains information on the following topics:

Feature description 929
Operating parameters 932
Feature interactions 932
Feature packaging 932
Feature implementation 932
Feature operation 934

Feature description

The United Kingdom Analogue Hardware Support feature provides the following capabilities:

- UK Analogue Trunk Enhancements
- UK Transmission Plans

UK Analogue Trunk Enhancements

Software changes have been implemented for the following hardware packs, in order to comply with UK standards:

- XDID (Extended DID trunk card)
- XCOT (Extended Central Office trunk card)

- XTD (Extended Tone Detector card), and
- XFEM (Extended Flexible E&M trunk card).

XDID

Situation	Solution
A DID trunk is not available for a new call.	A backward signal is sent to the Public Switched Telephone Network.
A short line and long line DID trunk requires support.	A 2dB Short Line (SHL) and Long Line (LOL) pad matrix have been defined.

XCOT

Situation	Solution
Support the following types of disconnect signaling required for Central Office trunks: • Earth Signaling (Ground Start), • Loop Calling (Disconnect Clearing), and • Loop Calling (Guarded Release) signaling.	The appropriate disconnect sequences have been programmed.
For Periodic Pulse Metering (PPM), an option is required to default to a meter pulse frequency of 50 Hz (the XCOT pack for the UK can only accept this value).	In the Route Data Block, if the PPM frequency is not prompted, the value will default to 50 Hz.
For Periodic Pulse Metering, the counting of buffered and unbuffered pulses.	The software has been modified to support both buffered and unbuffered PPM pulses.
A time-configurable detector is required to monitor the disconnection of loop trunks, disconnect clear trunks, and release guard trunks.	The Loop Calling Timer (LCT), with a configurable range of 128-32640 milliseconds, has been introduced in the Route Data Block.

Situation	Solution
UK ringing must be recognized.	To recognize UK ringing, the default value of the ring validation timer has been changed from 512 milliseconds to 256 milliseconds.
UK COT with Earth Signalling (Ground Start) or Loop Calling Disconnect Clearing provides hardware answer supervision.	The software has been modified to support answer supervision for both Earth Signalling (Ground Start) and Loop Calling Disconnect Clearing. Prompt SUPN appears for both types of signaling in LD 14. Answer supervision is not provided for Loop Calling Guarded Release.

XTD

The XTD pack can be configured, on a per-call basis, for either Dual-tone Multifrequency (DTMF) or Dial Tone Detection (DTD) signaling.

XFEM

The XFEM pack supports recorded announcement trunks, paging trunks, and music trunks, two-wire E&M, four-wire E&M, and 2280 Hz TIE trunks.

UK Digital Transmission Plans

Software changes have been implemented in order to comply with UK digital transmission plans for the following:

- Digital trunks, and
- Meridian Modular telephones.

Digital trunks

Situation	Solution
The transmission parameter values for digital trunks must be fixed.	The transmission parameter values for digital trunks are automatically downloaded, based on a zero default value.

Meridian Modular telephones

Situation	Solution
The transmission parameter values must be fixed and automatically downloaded, on a per-system basis.	The software has been changed to prevent transmission parameter prompts from appearing. The transmission parameters will be fixed for the UK, and will be downloaded on a per-system basis.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software..

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 13 – Configure the DTD/DTR data block..

2 LD 14 – Configure UK trunks, their associated signaling and transmission option.

3 LD 16 – Define the Loop Calling Detection Timer, and how the ground signal from a Recorded Announcement (RAN) machine should be interpreted for XFEM cards.

LD 13 – Configure the DTD/DTR data block.

Prompt	Response	Description
...		
TYPE	XTD	Extended Dial Tone Detector and Digitone Receiver data block.
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted when type = XTD. If a table other than 0 is entered, it must exist in LD 97.
- DTO	(NO) YES	Dial Tone Detection Only. (NO) = Do not disable DTR detection. YES = Disable DTR detection, only perform dial tone detection.

LD 14 – Configure UK trunks, their associated signaling and transmission option.

Prompt	Response	Description
...		
XTRK	XFEM XDID XCOT	Extended Flexible E&M trunk card. Extended DID trunk card. Extended CO trunk card.
...		
SIGL		Trunk signaling.
	LDC	Loop calling, disconnect clear. Accepted when TYPE = COT and UK package is equipped.
	LGR	Loop calling, guarded release. Accepted when TYPE = COT and UK package is equipped.
...		

CLS	(SHL) LOL NTC TRC VNL	(Short line) Long line Class of Service. Transmission Class of Service, where: NTC = Non-transmission Compensated TRC = Transmission Compensated, and VNL = Via Net Loss. For E&M4 Wire and AC15 defined on XFEM trunks, NTC is used for circuit switched network-to-PSTN Link connections, while VNL is used for circuit switched network-to-circuit switched network TIE connections. SHL replaces TRC and LOL replaces NTC and VNL for XDID and XCOT trunks in Phase 7C and later.

LD 16 – Define the Loop Calling Detection Timer, and how the ground signal from a Recorded Announcement (RAN) machine should be interpreted for XFEM cards.

Prompt	Response	Description
...		
TIMR	LCT 0-128-1280	Loop Calling Detection Timer in milliseconds. Default for COT trunks = 128. Default for all other trunks = 256.
...		
GRD		Determines how the ground signal from a RAN machine should be interpreted for XFEM cards.
	(PLAY)	The ground signal from the RAN machine indicates that the machine is playing.
	IDLE	The ground signal from the RAN machine indicates that the machine is idle.

Feature operation

No specific operating procedures are required to use this feature.

User Selectable Call Redirection

Contents

This section contains information on the following topics:

Feature description 935
Operating parameters 936
Feature interactions 937
Feature packaging 940
Feature implementation 941
Feature operation 943

Feature description

User Selectable Call Redirection (USCR), enhances the implementation of several existing features. First, it enables the user to modify DNs at the telephone for the following redirections:

- Flexible Call Forward No Answer DN (FDN)
- Hunt DN (HUNT)
- External Flexible Call Forward No Answer DN (EFD), and
- External Hunt DN (EHT).

The Station Control Password feature must be active, with passwords defined in LD 15, for the user to change these redirection DNs.

Second, it expands the number of selectable Ringing Cycle Options (RCOs) for Flexible Call Forward No Answer (CFNA) from one to three.

User assignment of redirection DNs

USCR permits the user to modify any of the redirection DN's for FDN, HUNT, EFD, and EHT from a rotary, push-button, or digital telephone.

Depending on the type of telephone, there are three ways to access this feature: using a Special Service Prefix Code (SPRE 9915), a Flexible Feature Code (FFC), or the User Selectable Redirection (USR) key.

The user can also change the RCO from a telephone after accessing USCR. For security reasons, the user must enter the Station Control Password (SCPW) before changing the redirection DNs or the RCO.

Ringing Cycle Options (RCOs) for CFNA

The original implementation of Call Forward No Answer provided a single option (CFNA in LD 15) that defined the number of normal ringing cycles before CFNA treatment. The value could be in the range of 1-15, with a default of 4. This value determined how many times the telephone rang before CFNA treatment was initiated.

The CFNA prompt is now replaced with prompts CFN0, CFN1, and CFN2, each of whose value can be in the range of 1-15, with a default of 4. The number of distinctive ringing cycles for CFNA is also expanded. The DFNA prompt in LD 15 is replaced with DFN0, DFN1, and DFN2, with the same value range and default.

Additionally, the Ringing Cycle Option (RCO) prompt appears in LDs 10 and 11 for each telephone. Its value, in the range of 0-2, is a pointer to the CFNx and DFNx entries in the Customer Data Block. The following chart explains the relationship of the RCO value and the CFNx and DFNx entries in the Customer Data Block.

Operating parameters

To assign or print the RCO for a telephone requires that it have the Flexible Call Forward No Answer Allowed (FNA) Class of Service or Message Waiting Allowed (MWA) Class of Service.

The user's telephone must have User Selectable Redirection Allowed (USRA) Class of Service and a Station Control Password (SCPW). The user must enter the correct password to access USCR.

Table 53
Relationship between RCO value and CFNx, DFNx contents

An RCO value (per telephone) of	Selects these CFNA and DFNA entries (with sample contents shown)	And has this effect
0	**CFN0 (Default value of 4)**	CFNA treatment after four rings
	DFN0 (Value set to 2)	CFNA treatment after two distinctive rings
1	**CFN1 (Value set to 6)**	CFNA treatment after six rings
	DFN1 (Value set to 5)	CFNA treatment after five distinctive rings
2	**CFN2 (Value set to 3)**	CFNA treatment after three rings
	DFN2 (Default value of 4)	CFNA treatment after four distinctive rings

Basic Rate Interface (BRI) telephones do not support USCR because they cannot access SPRE or FFC, and have no feature keys. Therefore, BRI telephones will always use the entries for CFN0 and DFN0.

The user cannot use USCR to initially configure call redirection features. The features must be equipped, and the initial call redirection DNs must be established, via a service change.

This feature cannot be used remotely. A user can only change redirection DNs or the RCO for the telephone being used to access USCR.

Feature interactions

Automatic Call Distribution

An Automatic Call Distribution (ACD) DN cannot be stored as a redirection DN unless the ACD queue is defined as a Message Center.

Attendant Administration

Attendant Administration does not support assigning the USR key, RCO, or USRA/USRD Class of Service.

Autodial

USCR does not support Autodial; Autodial cannot be used to dial all or part of the digits for USCR programming.

Call Forward All Calls

When CFW redirects a call from telephone A to telephone B, and telephone B does not answer, the RCO of telephone B determines how long it rings. After the designated number of rings, the FDN of telephone A redirects the call.

Call Forward by Call Type

USCR enables a user to assign EFD from the telephone.

Call Forward No Answer
Flexible Call Forward No Answer

The single parameters previously used to define normal ringing cycles (CFNA) and distinctive ringing cycles (DFNA) are expanded to three (CFN0-2 and DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific CFNA and DFNA entries for each telephone.

Call Forward No Answer, Second Level

The number of ringing cycles before Second Level Call Forward No Answer (SFA) is determined by the RCO for the ringing DN, as with CFNA.

Call Redirection by Time of Day

User Selectable Call Redirection is not supported by Call Redirection by Time of Day.

Dial Access to Features and Services

The 9915 feature code accesses USCR from an analog (500/2500 type) telephone or a Meridian 1 proprietary telephone. The user dials this code after dialing the SPRE.

Directory Number Delayed Ringing

With User Selectable Call Redirection (USCR) a user can change the number of CFNA/DFNA ringing cycles. If the user changes the CFNA/DFNA value so that CFNA takes place before the Directory Number Delayed Ringing timer runs out, none of the SCN/MCN keys will receive an audible notification.

Distinctive/New Distinctive Ringing

The single parameter previously used to define distinctive ringing cycles (DFNA) is expanded to three (DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific DFNA entry for each telephone.

DPNSS1 Diversion

The User Selectable Call Redirection feature triggers Diversion Validation. If the numbering plan is DPNSS1 then diversion occurs. Numbering plan routes are checked to determine if redirection DN's are through DPNSS1 on a first choice route basis. If the number plan is not a DN through DPNSS1, then User Selectable Call Redirection works as usual.

Enhanced Hot Line
Flexible Hot Line

An analog (500/2500 type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Hunting

User Selectable Call Redirection permits a user to change the HUNT DN or EHT from a telephone. An attendant DN is only allowed for HUNT and EHT if the customer has the attendant defined as a message center (LD 15 – MATT=YES).

Message Center (MC) and Message Waiting

USCR affects the number of times the DN rings before the call is forwarded to the Message Center. The RCO in the Terminal Number (TN) block of the Multiple Appearance Redirection Prime (MARP) for the called DN determines the number of times the DN rings.

Multiple Appearance Redirection Prime (MARP)

When a Multiple Appearance DN is rung, the determination of the number of ringing cycles for CFNA depends on the value of the MARP prompt in LD 17. If the value is "YES," the number of ringing cycles is determined by the RCO number of the DN that is classified as a MARP TN. If the DN is a Multiple Appearance DN (MADN), the RCO values in the other TN blocks for that DN are ignored.

If the MARP value is "NO," the RCO is taken from the first TN in the DN block with a primary appearance of the DN. If there is none, the last TN in the DN block is used.

Pretranslation

If Pretranslation (package 92) is enabled, the digits entered as the redirection DN are pretranslated before they are stored. Note that no Pretranslation occurs when the redirection DNs are used in such call processing features as Hunting or CFNA, eliminating the possibility that the redirection DN is pretranslated twice.

Short Hunting

USCR does not support changing the HUNT or EHT for a telephone with Short Hunt enabled. USCR also does not support entering "000" from a telephone as the HUNT.

Speed Call

Speed Call is not supported by USCR.

Feature packaging

Flexible Feature Codes (FFC) package 139 is a prerequisite for the user activation part of this feature because it provides for the Station Control Password.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 15 – Configure USCR in the Customer Data Block..

2 LD 10 – Configure USCR for analog (500/2500 type) telephones.

3 LD 11 – Configure USCR for Meridian 1 proprietary telephones.

4 LD 57 – Configure USCR Flexible Feature Codes.

LD 15 – Configure USCR in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	ADD, or change.
TYPE:	RDR	Call Redirection
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- CFN0	1-(4)-15	Number of normal rings for CFNA, Option 0.
- CFN1	1-(4)-15	Number of normal rings for CFNA, Option 1.
- CFN2	1-(4)-15	Number of normal rings for CFNA, Option 2.
- DFN0	1-(4)-15	Number of distinctive rings for DFNA, Option 0.
- DFN1	1-(4)-15	Number of distinctive rings for DFNA, Option 1.
- DFN2	1-(4)-15	Number of distinctive rings for DFNA, Option 2.
TYPE	FFC	Gate opener.
...		
- SCPL	(0-8	Length of Station Control Password. If 0 = password disabled, USCR cannot be used.

LD 10 – Configure USCR for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
RCO	(0) 1 2	Ringing Cycle Option for CFNA, in the range of 0-2, with a default of 0.
SCPW	xxx...xx	Station Control Password.
CLS	(USRD) USRA	User Selectable Redirection Class of Service (permitting SPRE and FFC access) (denied) allowed.
Note: The technician can use easy change to change the RCO and USRA/USRD CLS. At the ITEM prompt, type RCO <value> where the value is 0-2.		

LD 11 – Configure USCR for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
RCO	(0) 1 2	Ringing Cycle Option for CFNA, in the range of 0-2, with a default of 0.
SCPW	xxx...xx	Station Control Password.
CLS	(USRD) USRA	User Selectable Redirection Class of Service (permitting SPRE, FFC, and USR key access) (denied) allowed.
KEY	xx USR	Key number of the USR key.
Note: The technician can use easy change to change the RCO and USRA/USRD CLS. At the ITEM prompt, type RCO <value> where the value is 0-2.		

LD 57 – Configure USCR Flexible Feature Codes.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
CUST	xx	Customer number, as defined in LD 15
CODE	USCR ALL	Prompt for USCR FFC, or all FFC code types.
USCR	xxxxxxx yyyyyyy <CR>	USCR FFC (1-7 digits). Define additional FFC codes, as needed. Ends the entry of FFC codes.

Feature operation

As a prerequisite to accessing the feature, the conditions shown in Table 54 must be met for the selected access method.

Table 54
Requirements for accessing USCR

Requirement	Access Method		
	USR Key	SPRE	FFC
FFC package equipped	Yes	Yes	Yes
SCPL is defined (>0)	Yes	Yes	Yes
SCPW is defined	Yes	Yes	Yes
Telephone has USR key	Yes	No	No
USRA Class of Service defined	Yes	Yes	Yes
SPRE defined	No	Yes	Yes
USCR FFC defined	No	No	Yes

To assign/query a redirection DN using SPRE:

- Take the telephone off-hook, or press the DN key on a digital telephone.

- Enter the SPRE.
- Enter the USCR feature access code (9915).
- Enter the Station Control Password.
- Enter the USCR option code, as shown in Table 55.

Table 55
USCR option codes

Code	Used to assign
1	FDN redirection DN
2	HUNT redirection DN
3	EFD redirection DN
4	EHT redirection DN
5	RCO

- Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
- Place telephone on-hook, or press the Rls key on a Meridian 1 proprietary telephone.

To assign or query a redirection DN using the USR key:

- Press the dark USR key.
- Enter the Station Control Password.
- Enter the USCR option code from Table 55.
- Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
- Press the USR key again.

To assign or query a redirection DN using an FFC:

- Take the telephone off-hook, or press the DN key on a Meridian 1 proprietary telephone.

- Enter the USCR FFC.
- Enter the Station Control Password.
- Enter the USCR option code, as shown in Table 55.
- Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
- Place telephone on-hook, or press the **Rls** key on a Meridian 1 proprietary telephone.

Variable Flash Timing and Ground Button

Contents

This section contains information on the following topics:

Feature description 947
Operating parameters 948
Feature interactions 948
Feature packaging 949
Feature implementation 949
Feature operation 949

Feature description

These two methods of operation allow an analog (500/2500 type) telephone user to obtain special dial tone and activate various system features while on an established call. They are also used to return to the original call. Both of these functions are referred to as a recall. The following are the two parts of the feature.

Variable Flash Timing

This part is an enhancement to Flash Timing. It allows further flexibility in defining the limits for the flash. A minimum range of 20 to 768 milliseconds has been added and the maximum range has been extended to 1500 milliseconds. These settings are made on a customer basis in LD 15. A switchhook flash of less that the minimum is ignored and one of greater than the maximum is read as a disconnect. All flashes between the minimum and maximum provide a recall.

Ground Button

This part is an alternative to Flash Timing. It requires the installation of a ground button line card in place of a regular 500-type line card and analog (500/2500 type) telephones which have the ground button capability. The ground button can be depressed for any length of time over the minimum flash timing to provide a recall.

Operating parameters

Variable Flash Timing and Ground Button Operation are supported only on Digitone sets.

Ground Button Operation requires that a QPC532 Ground Button line card be installed on sets that have the capability, rather than a regular 500-type line card.

Using SPRE codes, it is possible to invoke the same features from an analog (500/2500 type) telephone as from a feature telephone.

Feature interactions

Message Waiting

The Ground Button Recall message to the software uses the same data store as the Message Waiting feature. The telephone state indicates which feature is active. The state is idle for Message Waiting and active for Ground Button Recall.

Feature packaging

This feature requires International Supplementary Features (SUPP) package 131.

Feature implementation

LD 15 – Assign minimum and maximum flash time for each customer member.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	TIM	Timers.
...		
- FLSH	xxx yyy	Minimum and maximum switchhook flash timer in milliseconds. xxx = 20-(45)-768. yyy = 384-(896)-1500. The timing specified will be used for EPE equipment only. XPE equipment will use the FLSH specified in LD 97.

Feature operation

Variable Flash Timing

Any switchhook flash between 20 and 1500 milliseconds provides a recall.

Ground Button

Pressing the ground button for any length of time over 20 milliseconds provides a recall.

Variable Guard Timing

Contents

This section contains information on the following topics:

Feature description 951
Operating parameters 951
Feature interactions 951
Feature packaging 952
Feature implementation 952
Feature operation 952

Feature description

The guard timing capability for a trunk prevents outgoing calls from reseizing trunks for a specified time after disconnection, thereby protecting trunks against glare conditions. This feature allows the customer to specify one guard timing interval for incoming call disconnection and one guard timing interval for outgoing call disconnection.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

LD 16 – Configure Guard Timers for trunk.

Prompt	Response	Description
...		
CFWR	(NO) YES	CFW restriction (not allowed) allowed.
IDOP	(NO) YES	Respond YES to allow the trunk CDR for internal calls to identify the originating station instead of the forwarding station.
TIMR	GTI 128-(896)-32640	Incoming Guard timer.
TIMR	GTO 128-(896)-32640	Outgoing Guard timer.

Feature operation

No specific operating procedures are required to use this feature.

Virtual Office

Contents

This section contains information on the following topics:

Feature description 953
Operating parameters 954
Feature interactions 955
Feature packaging 958
Feature implementation 958
Feature operation 958

Feature description

The Virtual Office feature enables users to login to any Internet telephone using their own User ID and password. Upon login, the Internet telephone de-registers with the Succession Call Server and registers to the Succession Call Server associated with the given User ID. This can be the same Succession Call Server or another Succession Call Server within the network. If it is another Succession Call Server, a Nortel Meridian Gatekeeper is required to provide the address of the Internet Telephone Terminal Proxy Server (TPS) node. Upon logout from Virtual Office, the Internet telephone registers to the Succession Call Server associated with the settings stored in its EEPROM: the S1 server and Terminal Number (TN).

In order to login using Virtual Office, the TN associated with the current Internet telephone registration must be configured with the class of service (CLS) VOLA (Virtual Office Login Allowed). The TN associated with the User ID for the login must be configured with the CLS VOUA (Virtual Office User Allowed). For more information on VOLA and VOUA, refer to LD 11 and LD 81 in *Software Input/Output: Administration* (553-3001-311).

The Virtual Office feature functions only with the i2002, i2004, and i2050 Internet telephones. Refer to the following documents for more information:

- *i2002 Internet Telephone User Guide*
- *i2004 Internet Telephone User Guide*
- *i2050 Software Phone User Guide*

The Virtual Office feature supports only one customer. If more than one customer is configured, the customer with the lowest customer number is supported.

Operating parameters

The minimum required software and hardware is:

- System software and hardware
- IPL 3.0
- i2050 software build 297
- Element management 1.0
- OTM release 2.0

Firmware upgrade

The Virtual Office feature requires UNIStim protocol version 2.5, which is supported in firmware version 1.33 or later. No firmware upgrade takes place during a Virtual Office login.

The umsUpgradeAll command has no impact on Internet telephones logged into Virtual Office. These telephones do not get reset. If the Virtual Office login is within the same Succession Call Server, the Internet telephone firmware is upgraded after the user has logged out. If the Virtual Office login is between different Succession Call Servers, the Internet telephone simply registers back to its home TPS, and follows the normal firmware upgrade rules for regular registration.

Feature interactions

Automatic Call Distribution (ACD)

Automatic Call Distribution (ACD) agents requiring the Virtual Office login, must have a separate, single appearance DN assigned to a programmable feature key. This unique DN is the User ID for Virtual Office login.

Branch Office DN

When a Virtual Office login is for a Branch Office DN, the Branch Office Internet telephone is preempted. There is one situation where a Branch Office DN becomes registered to both the Main Office and the Branch Office. It occurs when the Branch Office Internet telephone is in survival mode and there is a Virtual Office login for that DN. The situation is resolved when WAN connectivity resumes and the Internet telephone, in local mode, is redirected to register with the Main Office TPS in a logged out state.

Gatekeeper end points

Virtual Office support is limited to Internet telephones registered with Succession Call Servers that are configured as end points in the same Gatekeeper. Virtual Office login between Internet telephones registered with Succession Call Servers that are configured as end points of different Gatekeepers is not supported.

Internet telephone options

The Internet telephone options are retained within the telephone regardless of the registered User ID. Remote Virtual Office logins display the time and date of the remote Succession Call Server.

Multiple Appearance Directory Number (MADN)

Upon Virtual Office login using a Multiple Appearance Directory Number (MADN), the TN is selected to match the provided SCPW. If this is not desirable, then a separate, secondary, single appearance DN can be assigned. After login using the single appearance DN, the user receives the assigned keymap and feature key list including the MADN.

If there is no matching SCPW, an *Invalid ID* error displays. The login session assumes all the features of this TN (the same as a regular registration).

Optivity Telephony Manager (OTM)

The OTM End-User Station Administration includes the CLS VOLA/VOLD, and VOUA/VOUD from LD 11.

Phantom Terminal Numbers

Virtual Terminal Numbers and Phantom Terminal Numbers use the same TN range. The maximum number of Phantom TNs and Virtual TNs configurable on Small Systems and Succession 1000 systems is 640.

Registration lockout

If a user enters an incorrect password three times, the TN is locked from Virtual Office login for one hour. This is known as *Password guessing protection*. When locked, the error message SRPT0108 is written to the Succession Call Server history file and printed to the TTY. The craftsperson can unlock the TN by disabling and then enabling it in LD 32. This lockout does not survive re-registration of the Internet telephone.

Virtual Office login

When an Internet telephone performs a Virtual Office login to another Internet telephone on the same Succession Call Server, but registered with a different Signaling Server, or Media card, the preempted telephone remains logged out with the Logged Out screen, even after the login telephone has logged out. When the user returns to the home office, the Home softkey is used to log back into the home TN.

Virtual Office login using UDP

Virtual Office login to a Branch User using UDP from a Branch User in local mode always tries to log into the Branch User registered to the Main Office. Therefore, Virtual Office login using UDP will fail if the WAN connection to the Main Office is down.

Virtual Office preemption

A user who is already logged into Virtual Office can log in again on another Internet telephone. This is useful if the user forgets to log out of an Internet telephone and returns to his/her desk. The home Internet telephone or the logged in Virtual Office Internet telephone is preempted upon a successful login on another Internet telephone. The preempted Internet telephone remains registered to the TPS but not to the Succession Call Server, and is forced to register to the TN configured on the Internet telephone.

Since there can only exist one registration for a TN within a Succession Call Server, the Internet telephone is no longer operational, as indicated by the *Logout Screen* on the display.

The preempted Internet telephone can be used by another user for a Virtual Office login or it can be re-registered as the home Internet telephone. The preempted Succession Call Server registration may be a regular registration, a Virtual Office login, or a Branch User login. If it is a regular registration, then the subsequent login is resolved within the Succession Call Server to be a regular registration not a Virtual Office login or a Branch User login. A regular registration is one in which the Internet telephone's EEPROM S1 and TN settings match the login TN within the associated Succession Call Server.

If a virtually logged in Internet telephone resets, it attempts to register with the Succession Call Server associated with its S1 and TN. If the TN has a Virtual Office login or Branch User login, then it does not register to its Succession Call Server but displays the *Logout Screen*.

When attempting to login, if there is an existing login to the same User ID or a home Internet telephone for the User ID and it is not in the idle state (for example, ringing, ringback, or in a call) the new login attempt gets a *Login Busy* message.

Feature packaging

Virtual Office requires the following packages:

- Branch Office (SBO) package 390
- Virtual Office (VO) package 382
- Virtual Office (VO) package 387

Feature implementation

LD 11 – Configuring Virtual Office

Prompt	Response	Description
REQ:	NEW/CHG	
TYPE:	i2002 i2004 i2050	Internet telephones.
CUST	xx	Customer number, as defined in LD 15
BUID	<user id>	Dialable DN, Main Office User ID. Enter X to delete.
MOTN	 l s c u c u	Main Office Terminal Number For Large Systems For Small Systems and Succession 1000 systems
. . .		
CLS	(VOLA) VOLD	Allowed/deny Virtual Office login operation on this TN.
CLS	(VOUA) VOUD	Allow/deny Virtual Office user onto this TN using another telephone.

Feature operation

The Virtual Office feature supports the following operations:

- Virtual Office login

- Virtual Office connection
- Virtual Office logout

Virtual Office login

Since i2004 and i2050 Internet telephones have more key functions than the i2002 , a Virtual Office login from an i2002 Internet telephone to an i2004 or i2050 TN is blocked if:

- Key 0 is ACD.
- Any key (from key 1 to key 15) is defined as AAK, CWT, DIG, DPU, GPU, ICF, MCN, MCR, MSB, PVN, PVR, SCR or SCN.

If a login is attempted in the above situation, an error message is displayed.

Procedure 1
Performing Virtual Office login

This procedure instructs the user on how to log into Virtual Office. The Internet telephone has been configured using Branch User Config and is operating in Normal Mode.

1 Press the **Services** key to display the **Options** menu.

Note: The i2002 screen displays only one line at a time. Use the **up/ down arrow** key to scroll through the menu.

2 Use the **down arrow** key to highlight **Virtual Office Login**.

3 Press the **Select** softkey.

The screen prompts for the User ID.

4 Enter the User ID. This is the user's dialable DN with the access code.

Note: The User ID must be an ESN number. It can take the form of a UDP number, a CDP number, or a transferable DN.

5 Enter the Station Control Password (SCPW) for the Main Office Internet telephone.

When the User ID entered is not found locally the message, *Locating Remote Server*, displays indicating that connection setup is ongoing. A Nortel Meridian Gatekeeper provides the node IP address of the Home TPS associated with the User ID. The local TPS confirms connectivity to the remote TPS, then the Internet telephone is redirected to the remote Home TPS.

End of Procedure

Login failures and errors

A Virtual Office login fails if any of the following occurs:

1 When the login User ID is not local and there is a failure to get a response from a Gatekeeper.

2 When the User ID is not local and the Gatekeeper does not know the endpoint of the User ID.

3 When the User ID is not local and the Home TPS is unreachable.

4 When the User ID is not local and the Internet telephone firmware required by the remote TPS node software is incompatible with that provided by the local TPS node.

5 When the User ID is not known or a MADN cannot be resolved at the Home Succession Call Server.

6 When a SCPW is not configured for the User ID.

7 When the destination TN has Virtual Office User Denied (VOUD).

8 When there already exists a non-idle registered instance for the User ID.

9 When **Cancel** is selected by the user after a password failure.

10 When the password fails to authenticate after three attempts.

11 When the User ID entry in the Gatekeeper database points back to the originating Succession Call Server.

Refer to Table 1 for system messages, potential causes and available actions.

Table 1
System messages (Part 1 of 3)

Message	Cause	Action
Busy, try again	Remote Internet telephone is active (not idle).	Wait for remote Internet telephone to become idle and try again.
	ACD is logged in.	Logout ACD before Virtual Office login from another Internet telephone.
	Make-Set-Busy is inactive on ACD Internet telephone.	Set Make-Set-Busy active on ACD Internet telephone.
Invalid ID (1)	Incorrect User ID entered.	Enter correct User ID.
	User ID is not in Gatekeeper database.	Update Gatekeeper database to include User ID.
Invalid ID (2)	Incorrect User ID entered.	Enter correct User ID.
Invalid ID (3)	Incorrect User ID entered.	Enter correct User ID.
	User ID in Gatekeeper database points to originating Succession Call Server.	Change Gatekeeper configuration for the User ID to point to the correct endpoint.
Locked from Login	Three failed attempts to enter the correct SCPW.	Wait one hour for lock to clear automatically, or disable and enable the remote Internet telephone in LD 32 at the Succession Call Server to clear the lockout.
Logged Out	Home TN in use by Virtual Office.	Login to another TN using Virtual Office. Re-register as the Home (or Branch) Internet telephone.

Table 1
System messages (Part 2 of 3)

Message	Cause	Action
Permission Denied (1)	SCPW is not configured or enabled.	Configure a SCPW for the remote Internet telephone.
Permission Denied (3)	Incorrect User ID entered.	Enter correct User ID.
	SCPW is not configured.	Configure a SCPW for the remote Internet telephone.
Permission Denied (4)	Incorrect User ID entered.	Enter correct User ID.
	Attempt to login to an i2004/ i2050 TN from an i2002 Internet telephone.	Go to an i2004/i2050 Internet telephone and try again.
Permission Denied (5)	Incorrect User ID entered.	Enter correct User ID.
	The destination TN has Virtual Office User Denied (VOUD) configured.	Configure the remote Internet telephone with Virtual Office User Allowed (VOUA).
Permission Denied (6)	Incorrect User ID entered.	Enter correct User ID.
	Incorrect SCPW is entered.	Enter correct SCPW.
Required FW Vers	The User ID is not local and the firmware required by the remote TPS node software is incompatible with that provided by the local TPS node.	Update firmware.
Server Unreachable (1)	Gatekeeper is down.	Bring Gatekeeper up.
	The link to Gatekeeper is down.	Restore link to Gatekeeper.
Server Unreachable (2)	Remote TPS is down.	Bring remote TPS up.

Table 1
System messages (Part 3 of 3)

Message	Cause	Action
	Link to remote TPS is down.	Restore link to remote TPS.
	Remote Succession Call Server is down.	Bring remote Succession Call Server up.

Virtual Office connection

An Internet telephone is registered with the TN in its EEPROM and then a User ID and password are used to determine the home TPS for the Internet telephone during the Virtual Office connection. A Nortel Networks Succession Gatekeeper is required, if the home TPS is not the TPS where the Internet telephone is registered when the user initiates a Virtual Office login.

Upon login, the Internet telephone receives the features, time, date, and tones of the home Succession Call Server. The Internet telephone becomes part of the home zone and receives call service from this Succession Call Server. When this zone is remote the voice quality may be less than ideal because codec selection is based on the Internet telephone being in its expected location (subnet). Firmware upgrade on a TPS node does not upgrade the telephones logged into Virtual Office.

The Internet telephone options stored locally in the telephone's non-volatile memory retain the characteristics of the Internet telephone. That is, items from the Option menu such as language, display contrast, date and time format, and ring type retain the Internet telephone properties and do not change to reflect the preferences on the Virtual Office user's home Internet telephone.

If the Virtual Office user changes these settings, the changes persist in the Internet telephone even after logout, power-up, or reboot. In other words, changes to the Telephone Options menu items are retained (in the Internet telephone's EEPROM) regardless of who is registered. When changing any of these preferences, the original settings should be restored as a courtesy upon logout.

When connected to a Succession Call Server through Virtual Office, local trunks are the home CO and a call for emergency service is directed to the home PSTN. This is not ideal when the home is remote.

Virtual Office logout

The Virtual Office user initiates a logout by:

- performing a direct Virtual Office logout on the active Internet telephone (see Procedure 2 on page 965).
- returning to the home location and re-registering the home Internet telephone (using the **Home** softkey). This forces a logout on the idle Virtual Office Internet telephone registered to the same User ID.
- performing a Virtual Office login on another Internet telephone. This forces a logout on the idle Virtual Office Internet telephone registered to the same User ID.

The Virtual Office user is forced to logout when:

- the Internet telephone power cycles.
- the Internet telephone loses connectivity to the Home TPS.
- the automatic logout activates (see “Virtual Office auto-logout” on page 965).

Upon logout, the Internet telephone returns to its regular keymap unless the TN owner has an active Virtual Office login or Branch User login. In this case the Internet telephone displays the Logout screen. During a Virtual Office login, if connectivity to the Home Office is lost, calls are not maintained and the Internet telephone re-registers to its configured TN and Succession Call Server.

Procedure 2
Performing Virtual Office logout

This procedure instructs the user on how to logout of Virtual Office. The Internet telephone has been configured using Branch User Config and is operating in Normal Mode.

1 Press the **Services** key to display the **Options** menu.

Note: The i2002 screen displays only one line at a time. Use the **up/down arrow** keys to scroll through the menu.

2 Use the **down arrow** key to highlight **Virtual Office Logout**.

3 Press the **Select** softkey.

End of Procedure

Virtual Office auto-logout

The Virtual Office user can be configured to automatically logout all Virtual Office connections at a specified hour. This is configured in LD 15 using the following prompts:

- VO_ALO (YES/NO)–enables automatic logout.
- VO_ALOHR (0-23)–sets the hour to activate automatic logout.

Refer to *Software Input/Output: Administration* (553-3001-311) for more information on prompts and responses in LD 15.

Voice Call

Contents

This section contains information on the following topics:

Feature description 967
Operating parameters 968
Feature interactions 968
Feature packaging 970
Feature implementation 970
Feature operation 971

Feature description

Voice Call allows you to talk through the speaker of a Meridian digital telephone from another Meridian digital telephone. The called party does not have to lift the handset to hear you. For a two-way conversation, the called party must lift the handset or activate Handsfree, unless Handsfree Voice Call is enabled.

If the called telephone is busy on another DN, the caller hears continuous ringing. The called party hears a single beep and the Voice Call DN key flashes. If the telephone is busy on the Voice Call DN, the caller hears a busy tone. A fast busy tone may indicate that the Voice Call DN is no longer available (it may not be a Single Appearance DN).

Handsfree Voice Call

Handsfree Voice Call is a system feature that can be used with such telephones as the M2112, M2317, and M2616.

Handsfree Voice Call provides the option of configuring VCC/DIG (with voice option) to be answered in either Handsfree mode or loudspeaker only mode. Calls answered in Handsfree (HVA) mode establish a two-way voice path, while those answered in loudspeaker only (HVD) mode establish a one-way voice path from the calling telephone to the destination telephone.

Operating parameters

Both telephones must be Meridian digital telephones.

The Voice Call DN must be single appearance.

Handsfree Voice Call allowed/denied is set at the system level and can only be used with digital telephones that have Handsfree capabilities (such as the M2112, M2317, M2616). It requires Handsfree Allowed/HFA Class of Service on the destination telephone, which is set at the telephone level. Basic Rate Interface (BRI) and M3000 telephones do not support the Handsfree feature.

Feature interactions

Auto Answer Back

This feature is not affected by the Handsfree Voice Call feature.

Automatic Line Selection

This feature is not selected by automatic Outgoing Line Selection. It is selected for Incoming Ringing and Non-Ringing Line Selection.

Call Party Name Display

The telephone originating a Voice Call displays the called DNs Call Party Name Display. The called telephone shows the caller's DN and name on its display.

Display of Calling Party Denied

Display information on sets involved in a Voice call is based on the individual Class of Service of each set.

Flexible Feature Code Boss Secretarial Filtering

A call to a Voice Call key on a boss set with filtering active is not filtered to the secretary set.

Flexible Voice/Data Terminal Number

If a dynamic TN has a single appearance DN key that terminates on a Voice Call (VCC) key, the called party hears a single beep if occupied on another DN. However, if the called party is a dynamic TN in data mode, the DN key lamp flashes. A beep is not provided.

Hot Line

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line.

When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Manual Signaling

The same DN can be used for both Voice Call and Manual Signaling (Buzz) as long as it remains a Single Appearance DN.

Multiple Appearance DNs

If a Voice Call DN is added to a second telephone, the DN becomes a Multiple Appearance DN (MADN). Voice Call no longer works on that DN and fast busy tone is returned.

On Hold on Loudspeaker

It is possible to program this feature with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Feature packaging

This feature requires Voice Call requires the Optional Features (OPTF) package 1.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 11 – Configure Voice Call for the originating Meridian 1 proprietary telephone..

2 LD 15 – Configure Handsfree Voice Call for the system.

LD 11 – Configure Voice Call for the originating Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	aaaa	Telephone type, where: aaaa = SL1, 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, or 3000.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems
KEY	xx SCR yyy...y	Adds a single appearance single call key on the terminating telephone, where: xx = key number, and yyy...y = the DN assigned to the Voice Call key for the originating telephone.
KEY	xx VCC yyy...y	Adds a Voice Call key on the originating telephone, where: xx = key number, and yyy...y = the DN of the terminating telephone. This key activates the feature.

LD 15 – Configure Handsfree Voice Call for the system.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST	 0-99 0-31	Customer number For Large Systems For Small Systems and Succession 1000 systems
- OPT	(HVD) HVA	Handsfree Voice Call (denied) allowed.

Feature operation

Voice Call

To make a Voice Call:

- Lift the handset and press **Voice Call**. The DN is automatically dialed. If the called telephone is busy on another DN, you hear continuous ringing. If the telephone is busy on the Voice Call DN, you hear busy tone.

To end a Voice Call:

- Press **Rls**.

To answer a Voice Call on an idle telephone:

- Let the call ring once. The call is answered automatically, activating the Voice Call DN over the speaker. For a two-way conversation, lift the handset.

  If busy on another DN, you hear a single beep and the Voice Call DN flashes. You must end your present call to receive the Voice Call.

Handsfree Voice Call

HVA option

The originating telephone (telephone A) places a VCC/DIG call to the destination telephone (telephone B).

1 Telephone B rings once.

2 After one ring, telephone B automatically answers the call in Handsfree mode.

The DN and Handsfree LCDs are lit and a two-way voice path is established.

HVD option

Telephone A places a call to telephone B.

1 Telephone B rings once.

2 After one ring, telephone B automatically answers the call in loudspeaker only mode.

The DN LCD is lit and the Handsfree LCD remains dark, establishing a one-way voice path from telephone A to telephone B. At this point, telephone A is unable to hear the person at telephone B.

To reestablish a two-way voice path, telephone B must either go off-hook or press the Handsfree button.

Note: Busy calls are not changed by Handsfree Voice Call.

X08 to X11 Gateway

Contents

This section contains information on the following topics:

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Feature interactions 977
Feature packaging 978
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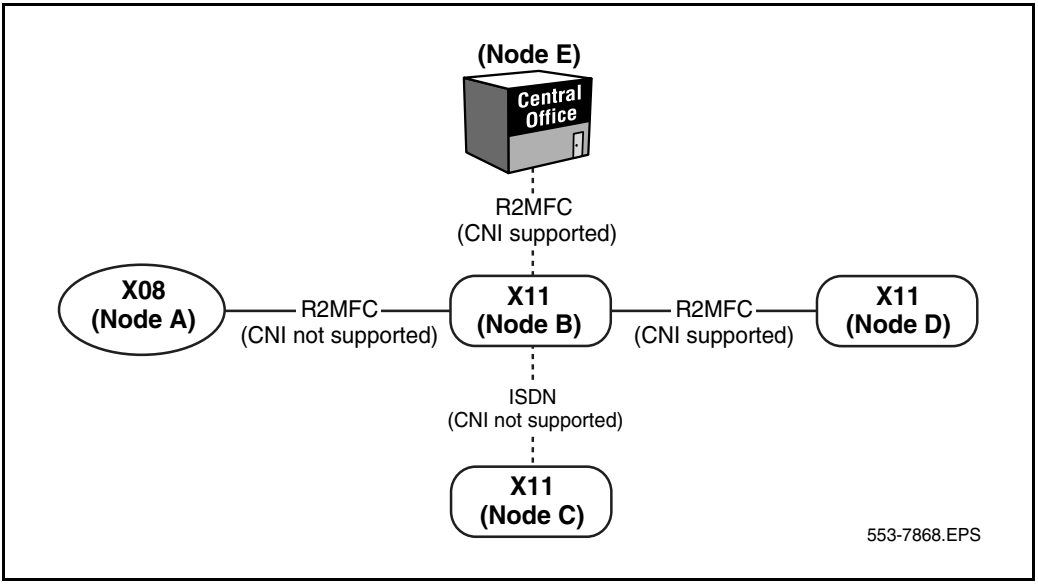
Feature description

X08/X11 Gateway is a feature which allows the use of both Generic X08 and Generic X11 software in the same network. This feature allows individual system nodes, running X08 and X11 software, to interface with one another. The Gateway makes this interconnection possible by allowing X11 nodes to “bridge” between both R2 Multifrequency Compelled (R2/MFC) signaling and L1 signaling, and Integrated Services Digital Network (ISDN) signaling. Although certain configurations of the X08 nodes may be necessary, no changes to X08 software are required.

This feature provides connectivity between X08 and X11 nodes, using L1, R2 Multifrequency Compelled (MFC), and Integrated Services Digital Network (ISDN) signaling protocols. The X08 L1 Signaling supports call setup, a numbering plan and Calling Number Identification (CNI). However, the L1 Signaling that is provided into X11 is a subset of the X08 L1 Signaling, supporting only the supplementary services required to support CNI and the suppression of Bring Up Receiver (BUR) signals.

Figure 24 summarizes the types of R2 MFC connections and tandems that are supported by the X08 to X11 Gateway.

Figure 24
R2 MFC Connections and Tandems



X08 to X11 connections using R2 MFC routes (Node A to Node B) – CNI is not supported because X08 does not provide outgoing CNI signaling.

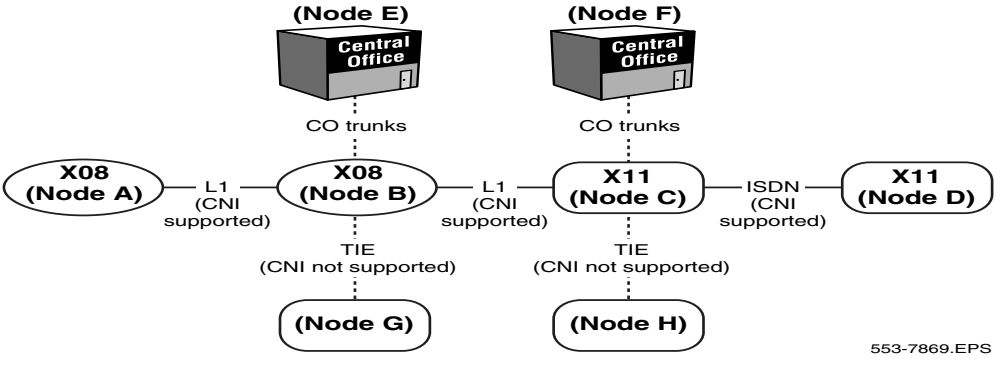
X11 to a private exchange using R2 MFC routes (Node B to Node E) – CNI is supported in both directions (DID/DOD).

Tandems using R2 MFC and ISDN routes, as follows:

- Tandems from an X08 node to an X11 node using R2 MFC routes to another X11 node using ISDN routes (Node A to Node B to Node C). CNI is not supported for this tandem.
- Tandems from one X11 node to another X11 node using R2 MFC routes to another X11 node using ISDN routes (Node D to Node B to Node C). CNI is not supported for this tandem.
- Tandems from a Public Exchange to an X11 node using R2 MFC CO routes to another X11 node using ISDN routes (Node E to Node B to Node C).

Figure 25 summarizes the types of L1 connections and tandems that are supported by the X08 to X11 Gateway (tandemning to X11 nodes using the R2 MFC Signaling is not allowed):

Figure 25
L1 MFC Connections and Tandems



X08 to X11 connections using L1 routes (Node B to Node C) – CNI is supported.

X08 or X11 connections to a private exchange using CO routes (Node B to Node E or Node C to Node F) – These routes can be analog or digital, and are non-R2MFC. CNI is not supported.

X08 or X11 connections to a private exchange using TIE routes (Node B to Node G or Node C to Node H) – These routes can be analog or digital, and are non-R2MFC. CNI is not supported.

Tandems using L1 and ISDN routes, as follows:

- Tandems from an X08 node to an X11 node using L1 routes to another X11 node using ISDN routes (Node B to Node C to Node D). CNI is supported.

- Tandems from an X08 node to an X11 node using L1 routes to a private exchange using analog or digital routes (Node B to Node C to Node F). CNI is not supported for this tandem.
- Tandems between X08 and X11 nodes using L1 routes to a node using TIE routes (Node B to Node C to Node H, and Node C to Node B to Node G). CNI is not supported.

Operating parameters

Routes using R2/MFC signaling can only be tandemed to routes using L1 signaling in cases where:

- the L1 route uses L1 Basic signaling (no supplementary services);
- the X11 node makes no Calling Number Identification (CNI) requests;
- the X08 node makes no call extensions for Ring Again (RGA); and
- signal assignment is co-ordinated between the X11 and X08 nodes.

L1 signaling in X11 must use TIE trunks.

L1-signaled routes will support CNI only when End-to-End Signaling is used.

The following groups of features do not operate on L1-signaled calls between X08 and X11 nodes:

- features requiring Bring Up Receiver (BUR) signals;
- call diversions;
- X08 trunk optimization;
- call transfer to an unestablished connection;
- Break-in, Recall, Incoming Call Identification (ICI) requests and Night Service Notification attendant features; and
- Ring Again (RGA).

X08 L1 signaling allows only one unsupervised trunk in a call connection. An X11 node tandeming an L1 connection from an X08 node does not inform the X08 node of unsupervised-trunk usage.

R2/MFC tandems support End-to-End Signaling only when the tandem node uses either the same R2/MFC table for both trunks or uses two tables with identical contents and the same End-to-End Signaling code. Calling Number Identification (CNI) is carried end-to -end even where End-to-End Signaling is not available.

X08 to X11 connections, using R2/MFC, do not support CNI. Outgoing CNI, on a tandem R2/MFC connection from an X08 node, uses the customer identifier of the tandeming X11 node, plus the Access Code of the route from the X08 node.

CNI is not supported over R2/MFC to ISDN tandem connections.

A third level of R2/MFC signaling, consisting only of backward signals, is not supported. This level of signaling is used for coin-box calls or calls from subscribers with home meters.

CNI in Call Detail Recording (CDR) records will have the same length only when all DNs, route access codes, trunk identifiers and attendant identifiers have the same length.

X08 does not have Integrated Services Digital Network (ISDN) capabilities.

Feature interactions

The network supported features using Gateway depends on the specific types of connection involved in any particular call.

Calling Number Identification (CNI)

Calling Number Identification (CNI) is supported on R2/MFC signaling connections between X11 nodes and Central Offices (COs), in both directions of calling, provided that the trunk being used has CNI-allowed Class of Service. CNI has the following characteristics across this type of connection:

- CNI begins with an optional customer identifier, 1-8 digits long;
- the customer identifier is followed by a caller identifier (a DN of 0-7 digits, an attendant identifier, a trunk identifier or a route access code);
- the attendant identifier has a maximum of 4 digits (identified on a customer basis); if the attendant identifier has not been defined, the attendant DN is used;
- the trunk identified has 0-7 digits (as assigned in LD 14);

- the trunk identifier does not have a unique value;
- the route access code is used if the trunk identifier has not been defined;
- a maximum of 16 digits of CNI can be carried across an R2/MFC connection, and
- end-to-end CNI to the CO works when the call tandems across more than one X11 node, using R2/MFC.

CNI is not supported on tandem connections between R2/MFC and ISDN routes.

In R2/MFC connections between X08 and X11 nodes, end-to-end CNI is only supported in cases where it has been requested by the X08 node. The X08 node will not support outgoing CNI. On an outgoing connection, the CNI supplied to the far end is that of the tandeming node when the tandeming node has not received CNI on an R2/MFC connection.

CNI is fully supported on R2/MFC connections between X11 nodes. End-to-end CNI to a CO works when the call tandems across more than one X11 node, using R2/MFC. If a tandeming node does not receive CNI, that node sends its own CNI forward.

CNI is supported on L1-signaled routes only when End-to-End Signaling is used.

Network Ring Again

Network Ring Again is not supported across any R2/MFC signaling connection or across L1-signaled connections between X08 and X11 nodes.

Feature packaging

The following packages are required for X08/X11 Gateway:

For R2/MFC Signaling, the following package is required:

- R2/MFC package 128

For L1 Signaling, the following packages are required:

- L1 package 188 and
- R2/MFC package 128.

For R2/MFC—ISDN Gateway, the following packages are required:

- Integrated Services Digital Network (ISDN) package 145;
- Primary Rate Access (PRA) package 146;
- DID-to-Network package 161; and
- R2/MFC package 128.

For L1—ISDN Gateway, the following packages are required:

- R2/MFC package 128;
- L1 package 188;
- Integrated Services Digital Network (ISDN) package 145;
- Primary Rate Access (PRA) package 146;
- DID-to-Network package 161; and
- Network Attendant Service (NAS) package 159 is required when the L1—ISDN Gateway must transport CNI.

Feature implementation

Task summary list

The following is a summary of the tasks in this section:

1 LD 14 – Assign trunk for X08 to X11 gateway.

2 LD 16 – Assign trunk route for X08 to X11 gateway.

LD 14 – Assign trunk for X08 to X11 gateway.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TIE	TIE trunk.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems and Succession 1000 systems

CUST	xx	Customer number, as defined in LD 15
TKID	nnnnnnn	Trunk Type Identifier (Does not have to be unique).

LD 16 – Assign trunk route for X08 to X11 gateway.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT	0-511 0-127	Route number For Large Systems For Small Systems and Succession 1000 systems
TKTP	TIE	TIE trunk type.
CCNI	(NO) YES	Call Number Indicator or CNI enabled on route.

Feature operation

No specific operating procedures are required to use this feature.

Meridian 1, Succession 1000,
Succession 1000M

Features and Services

Book 3 of 3 (N to Z)

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