Nortel Communication Server 1000 Features and Services Fundamentals— Book 4 of 6 (I to M)

Release 7.0
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Contents

Chapter 1: New in this release

This document has been updated with the following new features for Communication Server 1000 Release 7.0:

- IP Ad Hoc Conference on page 217
- IP Attendant Console on page 223
- IP Music on page 235
- IP Phone 1210 Last Number Redial soft key on page 259
- IP Phone Disable Mute function on page 267
- IP Phone Audio Message Waiting Indication on page 263
- IP Phone Password Protection for Language and Feature Key Labels on page 271
- IP Phone single-line-display of PD, CL, RL, and Corporate Directory additional information on page 275
- IP Recorded Announcements on page 279
- IP Tone on page 287

Others

This release contains no other changes.

Revision History

April 2013	Standard 04.10. This document has been up-issued to add more information to Zone Based Dialing in the Feature Interactions section of IP Network-wide Virtual Office on page 243.
September 2012	Standard 04.09. This document is up-issued to support Communication Server 1000 Release 7.0. A section for Zone Based Dialing has been added to the Feature interactions section of IP Network-wide Virtual Office on page 243.
May 2012	Standard 04.08. This document is up-issued to include the correct description for MAS fundamentals.

April 2012	Standard 04.07. This document is up-issued to include additional information on the Mobile Extension Handoff feature.
January 2012	Standard 04.06. This document is up-issued to support Communication Server 1000 Release 7.0. The IPE Blocking feature is deleted because it is removed during sysload in 7.0 and later releases.
July 2011	Standard 04.05. This document is up-issued to support Communication Server 1000 Release 7.0.
June 2011	Standard 04.04. This document is up-issued to support Communication Server 1000 Release 7.0.
May 2011	Standard 04.03. This document is up-issued to support Communication Server 1000 Release 7.0.
August 2010	Standard 04.02. This document is up-issued to support changes to the licensing details for IP Media Services applications.
June 2010	Standard 04.01. This document is up-issued to support Communication Server 1000 Release 7.0.
July 2009	Standard 03.04. This document is up-issued to support Communication Server 1000 Release 6.0.
June 2009	Standard 03.03. This document is up-issued to support Communication Server 1000 Release 6.0.
May 2009	Standard 03.02. This document is up-issued to support Communication Server 1000 Release 6.0.
May 2008	Standard 02.04. This document is up-issued for CS 1000 Release 5.5.
February 2008	Standard 02.03. This document is up-issued for CS 1000 Release 5.5.
December 2008	Standard 02.02. This document is up-issued for CS 1000 Release 5.5.
December 2007	Standard 02.01. This document is up-issued for CS 1000 Release 5.5.
July 2007	Standard 01.04. This document is up-issued for CR Q01692113 (revising the 500 Telephone Features and Bandwidth Management Support for Network Wide Virtual Office chapters in Book 1) and CR Q01673602 (revising the Conference Warning Tone Enhancement chapter in Book 2).
June 2007	Standard 01.03. This document is up-issued for CR Q01447763, revising the Software Licenses chapter in Book 6.

June 2007	Standard 01.02. This document is up-issued for CR Q01648906, revising the Network Music feature implementation in Book 5.
May 2007	Standard 01.01. This document is up-issued to support Communication Server 1000 Release 5.0. This document is renamed Features and Services Fundamentals (NN43001-106) and contains information previously contained in the following legacy documents, now retired: • Features and Services: Book 1 of 3 (A to C) (553-3001-306B1) • Features and Services: Book 2 of 3 (D to M) (553-3001-306B2) • Features and Services: Book 3 of 3 (N to Z) (553-3001-306B3)
July 2006	Standard 17.00. This document is up-issued to reflect the following changes: • Addition of M3900 Full Icon Support feature on pages 797 to 800 (Book 2), due to CR Q00926961. • Addition of M3900 Set-to-Set Messaging feature on pages 801 to 806 (Book 2), due to CR Q00926961. • Addition of M3900 series digital telephone feature reference on pages 341, 342 of the Personal Directory chapter (Book 3), due to CR Q00926961.
April 2006	Standard 16.00. This document is up-issued to reflect the following changes in content: • Addition of keycode commands for CP PIV on pages 595 to 610 (Book 2), due to CR Q01296486 • Addition of IPMG on CS1000E to the following: operating parameters on page 364 (Book 3); and LD 97 on page 379 (Book 3), due to CR Q01272524. • Additions to the following: Call Redirection by Day on page 848 (Book1); the CRDAY prompt on page 852 (Book 1); and Call Redirection by Time of Day on page 858 (Book 1), due to CR Q01297600. • Addition of Flexible Feature Codes to list on pages 371 to 376 of Flexible Feature Codes chapter (Book 2), due to CR Q01336199. • Correction to Message Intercept for Set Status Lockout on pages 982-983 (Book 2), due to CR Q01168852. • Correction to SECA001 alarm message on page 402 (Book 1), due to CR Q01223733. •

January 2006	Standard 15.00. This document is up-issued to reflect the following changes in content: • Addition of Converged Office feature on page 1247 (Book 1); changes to interactions with Call Forward All Calls on pages 647, 648, 721, 725 (Book 1), and 521 (Book 2), due to CR Q01200310. • Addition of IP Phones to supported sets referenced in Selectable Conferee Display and Disconnect on pages 667 to 700 (Book 3), due to CR Q01009956.
August 2005	Standard 14.00. This document is up-issued to support Communication Server 1000 Release 4.5.
September 2004	Standard 13.00. This document is up-issued for Communication Server 1000 Release 4.0.
October 2003	Standard 12.00. This document is issued for Succession 3.0.
November 2002	Standard 11.00. This document is up-issued to support Meridian 1 Release 25.40 and Succession Communication Server for Enterprise (CSE) 1000, Release 2.0. This is book 2 of a 3 book set.
January 2002	Standard 10.00. Up-issued to include content for Meridian 1 Release 25.40 and Succession Communication Server for Enterprise 1000, Release 1.1.
April 2000	Standard 9.00. This is a global document and is up-issued for Release 25.0x. Document changes include removal of: redundant content; references to equipment types except Options 11C, 51C, 61C, and 81C; and references to previous software releases.
June 1999	Issue 8.00 released as Standard for Generic Release 24.2x.
October 1997	Issue 7.00. This is the Release 23.0x standard version of this document. Certain application-specific features have been removed from this document and have been placed in their appropriate Nortel Networks technical publications (NTPs). Automatic Call Distribution features can be found in Automatic Call Distribution Feature description 553-2671- 110; Call Detail Recording features can be found in Call Detail Recording Description and formats 553-2631-100; Primary Rate Interface features can be found in International ISDN PRI Feature description and administration 553-2901-301; R2MFC and MFC features can be found in Multifrequency Compelled Signaling 553-2861-100; and DPNSS1 features can be found in DPNSS1 Features and Services 553- 3921-300.

August 1996	Issue 6.00. This is the Release 22.0x standard version of this document. The features Automatic Number Identification, Automatic Trunk Maintenance, Multi Tenant Service, Radio Paging and X08/11 Gateway have been incorporated into this document. Accordingly, the following Nortel Networks technical publications have been retired to reflect this change: 553-2611-200, 553-2751-104, 553-2831-100, 553-2721-111 and 553-2941-100.
December 1995	Issue 5.00. This is the Release 21.1x standard version of this document.
July 1995	Issue 4.00. This is the Release 21 standard version of this document.
October 1994	Issue 2.0. This is the Release 20.1x soak version of the document.
July 1994	Issue 1.0. This is the Release 20.0x standard version of this document.

Chapter 2: How to get help

This chapter explains how to get help for Nortel products and services.

Getting help from the Nortel web site

The best way to get technical support for Nortel products is from the Nortel Technical Support web site: http://www.nortel.com/support.

This site provides quick access to software, documentation, bulletins, and tools to address issues with Nortel products. From this site, you can:

- download software, documentation, and product bulletins
- search the Technical Support Web site and the Nortel Knowledge Base for answers to technical issues
- sign up for automatic notification of new software and documentation for Nortel equipment
- open and manage technical support cases

Getting help over the telephone from a Nortel Solutions Center

If you do not find the information you require on the Nortel Technical Support web site, and you have a Nortel support contract, you can also get help over the telephone from a Nortel Solutions Center.

In North America, call 1-800-4NORTEL (1-800-466-7835).

Outside North America, go to the following web site to obtain the telephone number for your region: http://www.nortel.com/callus.

Getting help from a specialist by using an Express Routing Code

To access some Nortel Technical Solutions Centers, you can use an Express Routing Code (ERC) to quickly route your call to a specialist in your Nortel product or service. To locate the ERC for your product or service, go to: http://www.nortel.com/erc.

Getting help through a Nortel distributor or reseller

If you purchased a service contract for your Nortel product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller.

Chapter 3: Features and Software options

Package Name	Number	Mnemonic	Release
1.5 Mbit Digital Trunk Interface • Hong Kong Digital Trunk Interface • Reference Clock Switching (See also packages 129, 131, and 154)	75	PBXI	5
16-Button Digitone/Multifrequency Telephone • 16-Button Digitone/Multifrequency Operation	144	ABCD	14
2 Mbit Digital Trunk Interface • 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	129	DTI2	10

Package Name	Number	Mnemonic	Release
- 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)			
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			

Package Name	Number	Mnemonic	Release
Analog Calling Line Identification	349	ACLI	25
• CLID on Analog Trunks for Hong Kong (A-CLID)			
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			
Attendant Forward No Answer	134	AFNA	14
• Attendant Forward No Answer			
• Attendant Forward No Answer Expansion			

Package Name	Number	Mnemonic	Release
• 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			
• 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)			

Package Name	Number	Mnemonic	Release
• 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			
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

Package Name	Number	Mnemonic	Release
• 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			
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			

Package Name	Number	Mnemonic	Release
• 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			
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)			

Package Name	Number	Mnemonic	Release
Basic Routing	14	BRTE	1
• Basic Routing			
Boss Secretary Filtering (FFC activation)	198	FTCSF	15
• Flexible Feature Code Boss Secretarial Filtering			
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 Network Express	367	BNE	25
• Business Network Express/EuroISDN Call Diversion			
• Business Network Express/EuroISDN Explicit Call Transfer			
• Business Network 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			
Call Detail Recording, Data Link	6	CLNK	1

Package Name	Number	Mnemonic	Release
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			
Call Park Networkwide	306	CPRKNET	22
• Call Park Network Wide			
Call Park	33	CPRK	2
• Call Park			
• Recall after Parking			
• Call Pickup			

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

Package Name	Number	Mnemonic	Release
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)			
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

Package Name	Number	Mnemonic	Release
• Commonwealth of Independent States Digital Trunk Interface			
• Three-Wire Analog Trunk – CIS			
• Commonwealth of Independent States Automatic Number Identification (ANI) Digits Manipulation and Gateways Enhancements			24
• 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			
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			

Package Name	Number	Mnemonic	Release
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			
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)			
Digital Private Network Signaling Network Services (DPNSS1)	231	DNWK	16
• Attendant Call Offer			
• Attendant Timed Reminder Recall and Attendant Third Party Service			
• Call Back when Free and Next Used			

Package Name	Number	Mnemonic	Release
• 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			
Digital Private Network Signaling System 1 Message Waiting Indication	325	DMWI	23
• DPNSS1 Message Waiting Indication			
Digital Private Network Signaling System 1	123	DPNSS	16
• 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			
Direct Inward Dialing to TIE (Japan only)	176	DTOT	16
• Direct Inward Dialing to TIE			
• 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			

Package Name	Number	Mnemonic	Release
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			
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)			

Package Name	Number	Mnemonic	Release
• 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)			
Enhanced Controlled Class of Service	173	ECCS	15
Enhanced DPNSS Services	288	DPNSS_ES	21
• DPNSS1 Executive Intrusion			
Enhanced DPNSS1 Gateway	284	DPNSS189 I	20
• Enhanced DPNSS1 Gateway			
Enhanced Hot Line	70	HOT	4/10
• Hot Line			
• Network Intercom			
• Enhanced input/output buffering			
• Enhanced Maintenance (Patching)			
Enhanced Music	119	EMUS	12
• Music, Enhanced			
Enhanced Night Service	133	ENS	20
• Enhanced Night Service			
• Enhanced package printout			

Package Name	Number	Mnemonic	Release
• Equal Access Compliance			
Euro ISDN Trunk - Network Side	309	MASTER	22
• EuroISDN Trunk - Network Side			
Euro ISDN	261	EURO	20
• ISDN – Advice of Charge for EuroISDN			
• ISDN BRI and PRI Trunk Access for Europe (EuroISDN)			
• EUROISDN Continuation			
Euro Supplementary Service	323	ETSI_SS	22
• EuroISDN Call Completion Supplementary Service			
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			
Flexible Feature Codes	139	FFC	15

Package Name	Number	Mnemonic	Release
• 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 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			
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			
Geographic Redundancy Primary system	404	GRPRIM	4.0
Geographic Redundancy Secondary system	405	GRSEC	4.0
Group Call	48	GRP	1

Package Name	Number	Mnemonic	Release
• 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			
• Branch Office			
HiMail Fax Server	195	FAXS	18
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			

Package Name	Number	Mnemonic	Release
- 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			
Integrated Digital Access	122	IDA	16
• 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			
Integrated Message System UST and UMG are part of IMS Package.	35	IMS	2
• Integrated Messaging System Link			
Integrated Services Digital Network Application Module Link for Third Party Vendors	153	IAP3P	13
• Application Module Link			
• Network Application Protocol Link Enhancement			
Integrated Services Digital Network BRI Trunk Access	233	BRIT	18
• Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 235)			
Integrated Services Digital Network Supplementary Features	161	ISDN INTLSUP	14
• Call Connection Restriction (see also packages 146 and 147)			
• Direct Inward Dialing to Network Calling			
• Incoming Digit Conversion Enhancement			

Package Name	Number	Mnemonic	Release
• Network Time Synchronization			
• X08 to X11 Gateway			
Integrated Services Digital Network Signaling Link	147	ISL	13
• Call Connection Restriction (see also packages 146 and 161)			
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 Interactions, 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			
IP Media Gateway	403	IPMG	4.0
ISDN Semi-Permanent Connection	313	ISPC	22
• ISDN Semi-Permanent Connections for Australia			
• Italian Central Office Special Services (see also packages 129, and 157)			
Japan Central Office Trunks	97	JPN	9
• Japan Central Office Trunk			
Japan Digital Multiplex Interface	136	JDMI	14
• Japan Digital Multiplex Interface			

Package Name	Number	Mnemonic	Release
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			
• LOGIVOX Telephone			
• Loop Start Answer Supervision XUT			
• Loop Start Supervisory Trunks			
• Loop Start Supervisory Trunks (Incoming Calls)			
Location Code Expansion	400	LOCX	4.0
M2000 Digital Sets	88	DSET	7

Package Name	Number	Mnemonic	Release
• Distinctive Ringing for Digital Telephones • M2317 Telephones • Flexible Voice/Data Terminal Number			
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_RG A_ 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			
Make Set Busy • Make Set Busy • Make Set Busy Improvement	17	MSB	1
• Malicious Call Trace on Direct Inward Dialing			
Malicious Call Trace • Enhanced Malicious Call Trace • Malicious Call Trace • Malicious Call Trace DN/TN Print • Malicious Call Trace Idle	107	MCT	10
• Manual Line Service			

Package Name	Number	Mnemonic	Release
• 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
• Nortel Networks Integrated DECT			
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
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			

Package Name	Number	Mnemonic	Release
• 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			
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
Mobile X	412	MOBX	5.5
Mobility Server	301	MOSR	22
• Modular Telephone Relocation			
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)			

Package Name	Number	Mnemonic	Release
• 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			
• 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			

Package Name	Number	Mnemonic	Release
Multi-purpose Serial Data Link Single Terminal Access • Single Terminal Access	228	MSDL STA	19
Multi-purpose Serial Data Link • Multi-purpose Serial Data Link	222	MSDL	18
Multi-Site Mobility Networking	370	MSMN	25
Multi-User Login • Multi-User Login	242	MULTI_USER	19
Music Broadcast • Music Broadcast	328	MUSBRD	23
Music • Music	44	MUS	1
N/W Communications Management Center	30	CMAC	1
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

Package Name	Number	Mnemonic	Release
Network Class Of Service	32	NCOS	1
• Network Class of Service			
Network Message Services	175	NMS	16
Network Priority Queuing	60	PQUE	1
• Network Priority Queuing			
Network Signaling	37	NSIG	2
• Network Signaling			
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			

Package Name	Number	Mnemonic	Release
Nortel Symposium Call Center	311	NGCC	22
North America National ISDN Class II Equipment	291	NI2	21
• 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
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			

Package Name	Number	Mnemonic	Release
• 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)			
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			

Package Name	Number	Mnemonic	Release
• Periodic Camp-on Tone			
• Periodic Clearing			
• Periodic Clearing Enhancement			
• Periodic Clearing on RAN, ACD, Meridian Mail, 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			
• Privacy Release			
• Private Line Service			
Proactive Voice Quality Management	401	PVQM	4.0

Package Name	Number	Mnemonic	Release
Property Management System Interface	103	PMSI	10
• Property Management System Interface			
• Public Switched Data Service			
Pulsed E&M (Indonesia, French Colise)	232	PEMD	18
• Pulsed E&M DTI2 Signaling			
Q Reference Signaling Point Interface	263	QSIG	20
• Integrated Services Digital Network QSIG Basic Call			
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 with Priority during Night Service			
• Recall With Priority during Night Service			
• Recall With Priority during Night Service Network Wide			
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			

Package Name	Number	Mnemonic	Release
• Reference Clock Switching			
• Reference Clock Switching (see also packages 75, 129, and 154)			
Remote IPE • Remote Intelligent Peripheral Equipment	286	REMOTE_IPE	
Remote Virtual Queuing • Network Drop Back Busy and Off-hook Queuing (see also package 62) • Remote Virtual Queuing	192	RVQ	18
Resident Debug	82	RSDB	9
• Restricted Call Transfer			
• Ring and Hold Lamp Status			
• Ringback Tone from Meridian 1 Enhancement			
Ringing Change Key • Ringing Change Key	193	RCK	15
Room Status • Room Status	100	RMS	10
Scheduled Access Restrictions • Scheduled Access Restrictions	162	SAR	20
• Secrecy Enhancement			
• Secretarial Filtering			
• Seizure Acknowledgment			
• Selectable Conferee Display and Disconnect			
• Selectable Directory Number Size			
Semi-Automatic Camp-On • Attendant Blocking of Directory Number • Attendant Idle Extension Notification • Semi-Automatic Camp-On	181	SACP	15
• Serial Port Expansion			

Package Name	Number	Mnemonic	Release
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			
Single Term Access	228	STA	19
• Single Term Access			
• Slow Answer Recall Enhancement			
• Slow Answer Recall for Transferred External Trunks			
• Source Included when Attendant Dials			
SIP Gateway and Converged Desktop	406	SIP	4.0
Soft Switch	402	SOFTSWIT CH	4.0
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			

Package Name	Number	Mnemonic	Release
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			
• 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)			
• 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			

Package Name	Number	Mnemonic	Release
• Tone to Last Party			
• Tones, Flexible Incoming			
Traffic Monitoring	168	TMON	
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			
Virtual Network Services	183	VNS	16

Package Name	Number	Mnemonic	Release
• Virtual Network Services			
• Virtual Network Services/Virtual Directory Number Expansion (see also package 58)			
• Voice Call			
Virtual Office	382	VIRTUAL_ OFFICE	25
• Branch Office			
• Emergency Services For Virtual Office			
• Internet Telephone Virtual Office			
• Virtual Office			
Virtual Office Enhancement	387	VOE	3.0
• Branch Office			
• Emergency Services For Virtual Office			
• Internet Telephone Virtual Office			
Voice Mailbox Administration	246	VMBA	19
• Meridian Mail Voice Mailbox Administration			
X08 to X11 Gateway	188	L1MF	15
• X08 to X11 Gateway			
Zone Call Admission Control	407	ZCAC	4.5
• Adaptive Network Bandwidth Management			

Chapter 4: Hunting Trunk

Contents

This section contains information on the following topics:

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

Table 1: 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		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
SRCH	(LIN) RRB	Linear or Round Robin Hunting.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 5: ICP Network Screen Activation, Flexible DN Meridian Mail Interactions

Contents

This section contains information on the following topics:

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 telephone, the call is established and presented on the ICT at the terminating ICP position telephone.

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)

Because 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, because 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
- 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, Hunt

When a call redirected by Call Forward All Calls, Call Forward No Answer, Call Forward Busy, or Hunt terminates on an 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 (ESN) functionality which is supported is Coordinated Dialing Plan (CDP).

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 telephone with call forward active, the slow answer recall timer at the originating node is reset for ICP forward.

Slow Answer Recall for External Transferred Calls

When an ICP position telephone transfers an external call across an ISDN network, the slow answer recall timer is set at the transferring node to prevent the terminating telephone 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. Table 2: LD 15 on page 89

   For NWSA, enter up to 13 digits for ICDN and ECDN.
2. Table 3: LD 15 on page 90

   For FXDN, set the DN length and any padding digits at the ICDL and ICPD prompts.
3. Table 4: LD 15 on page 90

   For MMIA, enter YES to both the Meridian Mail prompt and the Intercept Computer prompt.

Table 2: LD 15

Prompt	Response	Description
...		
TYPE:	ICP	Intercept computer update.
- ICP	YES	ICP is available.

Prompt	Response	Description
...		
- ICMM	0-9	Message number.
- ICDN	0-13	Default internal DN.
- ECDN	0-13	Default external DN.

Table 3: LD 15

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.

Table 4: LD 15

Prompt	Response	Description
...		
TYPE:	IMS:	Integrated Message Service Options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
...		
IMS	YES	Meridian Mail is available for customer number.
...	...	...
TYPE	ICP–DATA	Intercept computer update.
- ICP	YES	ICP is available for customer number.
...		
TYPE:	ICP	Intercept computer update.
- ICP	YES	ICP is available.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 6: In-Band Automatic Number Identification

Contents

This section contains information on the following topics:

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 Fundamentals., NN43001-550*.

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 is 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 Fundamentals, NN43001-551*.

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 telephone display.

ACD Interflow

If an IANI call interflows to another predesignated local ACD DN, the ANI number is displayed on the overflow agent 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 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 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 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 telephone 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 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. Table 5: LD 16 on page 98

   Identify the route as an In-Band Automatic Number Identification route.
2. Table 6: LD 23 on page 99

   Send the IANI messages across the Auxiliary Processor Link (APL).

Table 5: LD 16

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.

Prompt	Response	Description
IANI	(NO) YES	(Disable) enable the IANI route.

Table 6: LD 23

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.

Chapter 7: Incoming Call Indicator Enhancement

Contents

This section contains information on the following topics:

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

Table 7: 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.

Chapter 8: Incoming DID Digit Conversion

Contents

This section contains information on the following topics:

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

- 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

- 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

- 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

- 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. Because 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 telephone, not necessarily the Direct Inward Dialing (DID) number to dial to reach the telephone. 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 telephone, not necessarily the DID number to dial to reach the telephone. 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. Table 8: LD 15 on page 110

   Specify the maximum number of Incoming Digit Conversion trees allowed.
2. Table 9: LD 49 on page 110

   Create the IDC tables to convert incoming Direct Inward Dialing digits by specifying the IDC tree and customer numbers.
3. Table 10: LD 16 on page 110

   Enable the digit conversion for required DID trunk routes.

Table 8: LD 15

Prompt	Response	Description
REQ	CHG	Change
TYPE	FCR	New Flexible Code Restrictions Option
CUST		Customer number
	0-99	Range for Large System and CS 1000E system
	0-31	Range for Small System and Media Gateway 1000B
- 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 The sum of the values for MAXT and DCMX cannot exceed 255 per customer.

Table 9: LD 49

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

Table 10: LD 16

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route Data Block
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system
	0-127	Range for Small System and Media Gateway 1000B

Prompt	Response	Description
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.

Chapter 9: Incoming Digit Conversion Enhancement

Contents

This section contains information on the following topics:

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. Because 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 telephone, not necessarily the Direct Inward Dialing (DID) number to dial to reach the telephone. 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 telephone, not necessarily the DID number to dial to reach the telephone. 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 105 in this document.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 10: Information Notification Service for Japan

Contents

This section contains information on the following topics:

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 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
  - 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 1: System with INS-J feature operating on page 119 shows the operation of the INS-J feature.

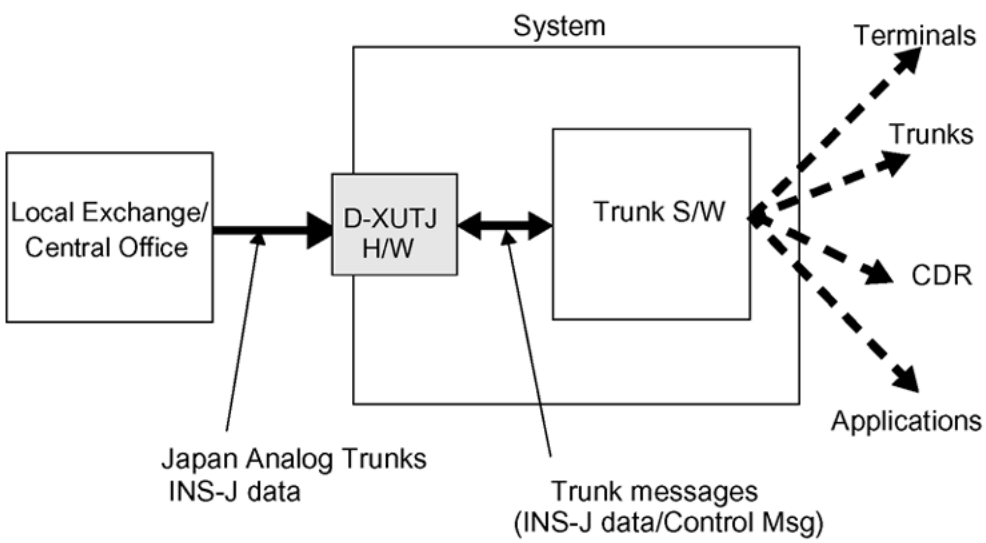


Figure 1: System with INS-J feature operating

Figure 2: System composition for INS-J CLID delivery on page 120 shows the system composition required for the INS-J CLID delivery.

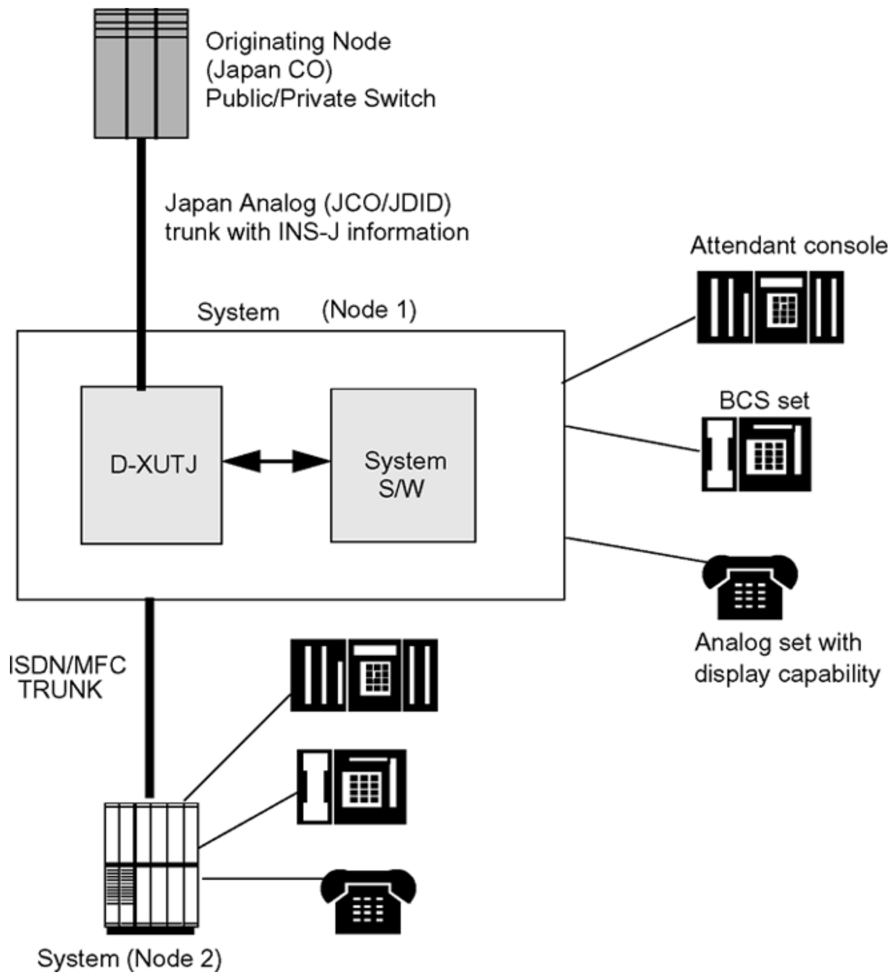


Figure 2: System composition for INS-J CLID delivery

Operating parameters

This feature is only applicable for incoming analog trunks. If the terminating telephone/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 are processed.

Display of Katakana characters is not supported. Any Katakana characters received are 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, because it is not an established call. In the case of an established call, the call is 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 telephone.

Call Transfer/Blind Transfer

When a telephone completes a Transfer/Blind Transfer of an incoming INS-J call, the Analog CLI information is delivered to the terminating telephone.

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 telephone.

CLASS

If the call terminates on a CLASS telephone then the Analog CLI information is passed to the CLASS feature.

Conference/No Hold Conference

When a telephone receives an incoming call and then initiates a conference call, the information of the initiating telephone is delivered to the terminating telephone, 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 telephone.

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. Because 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. Table 11: LD 14 on page 123

   Configure the Analog CLI Class of Service on a port-by-port basis

2. Table 12: LD 16 on page 123

   Configure the new ring validation timer

Table 11: LD 14

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 BTS	Japan CO or Japan DID Busy Tone Supervision (Optional)
CLS	(CLID) CLIA	Calling Line Identification denied or allowed.
...	...	

Table 12: LD 16

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

Prompt	Response	Description
...	...	
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.

Chapter 11: Instant License

Contents

This section contains information on the following topics:

Feature description

Use the Instant License feature (formerly called Instant ISM) to upgrade the License limits on your system. The License limits determine the maximum number of TNs, ACD positions, and other parameters on the system. With Instant License, you can deliver the necessary upgrade keycodes to the system without the need for a Sysload.

To activate keycodes, use the existing prompts in LD 143. If the upgrade keycode is eligible for instant activation, then the License limits upgrade “instantly”. After successful activation, a system message introduced by the Instant License feature indicates the following: that the keycode is accepted, License limits are increased, and a Sysload is not required.

Keycodes qualify for instant activation when the only difference between the upgrade keycode and the current system keycode is an increase in the number of License limits. Any decrease in License limits renders the keycode ineligible for instant activation. Likewise, changes made to other keycode parameters (including the addition or removal of feature packages, software release and issue, software generic (system type and call processor type), or AUX-ID) also render the keycode ineligible.

If a keycode is ineligible for instant activation (License parameters are lowered, or software packages are changed), then the following system message displays: “CCBR009 New keycode accepted. It is activated during the next restart”. In this case, you must perform a Sysload to complete keycode activation.

Operating parameters

The system does not treat the Small SystemMOPT parameter as a License limit, but rather as a package. The Instant License feature does not support instant MOPT changes. If the MOPT parameter is changed, a Sysload is required.

System initialization

If system initialization occurs during the activation of a new keycode, the system software attempts to complete the keycode activation. However, depending on when initialization begins, the software may not be able to complete keycode activation.

After the system completes initialization, use LD 22 to print the active License parameters. If keycode activation is successful, the printed License parameters match the new keycode parameters. If the printed License parameters do not match the new keycode parameters, but instead show the pre-upgrade parameters, then the system administrator must perform one of the following actions, depending on the system:

- For Large Systems, use the **KSHO** command in LD 143 to show the contents of the currently used keycode file.

  CP PII systems store keycodes on the hard disk or on floppy disk. CP PIV systems store keycodes on the Fixed Media Device (FMD) or on the Removable Media Device (RMD). For CP PII, use the “KSHO HD” command to verify keycodes on the hard drive. For CP PIV, use “KSHO FMD” to verify keycodes on the FMD.

  If the KSHO command shows the new keycode on the hard drive or the FMD, use the “KOUT” command to remove the pending keycode file, then restart the keycode activation processs. If the new keycode does not show on the hard drive or the FDM, then the administrator must perform a new keycode installation, using LD 143.
- For Small Systems access LD 143 and perform the system upgrade process again.

Feature interactions

License

Instant License does not change the operation of the various License limits. Instant License allows the user to upgrade License limits without having to Sysload.

IS-41 Networking

Instant License supports the MOB License parameter in the IS-41 Networking feature.

RAN and Music Broadcast

Certain traffic reports peg the number of times the RAN and Music License limits had been reached. With the Instant License feature, License limits may change instantly (without a Sysload). When this happens, a traffic report that is counting the License hits over a specified time period is checking against two different values consecutively. Therefore, for that time period, the report has an aberration.

Electronic Brandlining (EBLN/BRAND)

Unlike other License parameters which define the maximum configuration limits for various resources, the BRAND License parameter defines which Electronic Brandlining feature option the system is allowed to use.

The same limitation applies to the BRAND parameter as applies to other License limits, that is, the BRAND parameter must be unchanged or increased if the License limits are to be updated instantly without the need for a Sysload.

Once the BRAND License parameter has been increased, the user must access LD 17, to configure the actual string that is to be displayed, (as in the existing operation).

Telephone displays that display brandline information (when in an idle state) do not have the brandline updated immediately. The update occurs on a telephone the next time the LAMPAUDIT routine audits the telephone.

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 has three component parts:

1. Instant License parameter upgrade using a keycode diskette (CP PII systems) or a Removable Media Device (CP PIV systems)
2. Instant License parameter upgrade using HyperTerminal®
3. Instant License parameter upgrade for Small Systems

Instant License parameter upgrade using a keycode diskette (CP PII systems) or a Removable Media Device (CP PIV 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. Log in on a system terminal and access LD 143.

   `>LD 143`

   `>LD 143 CCBR000 .`

2. For CP PII systems, insert the new keycode diskette into the floppy drive on the active core. For CP PIV systems, insert the Removable Media Device (RMD) into the Compact Flash (CF) socket on the faceplate.
3. Enter the KDIF command and select keycode comparison options.

   Ensure that the new keycode does not lower License limits or reduce features compared with the existing keycode. If you have determined that the keycode lowers

License limits or reduces features, do not continue with the KNEW command, but contact your Nortel order management representative.

.KDIF Use **KDIF <param1> <param2>**with the following parameters

NEW	accepted new keycode
REC	currently used keycode
OLD	previously used keycode
CP PII	
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
CP PIV	
RMD	candidate keycode on Removable Media Device
FMD	candidate keycode created manually (KMAN) or uploaded (KUPL)

Enter the keycode comparison option. The new keycode option is shown in bold.

The following example compares the currently used keycode (REC) with the new keycode on the disk in floppy drive F0 (CP PII systems) or on the RMD (CP PIV systems). The limits shown are for example purposes only.

For CP PII: **.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	:	

( ) indicates that information is not available.

License 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 is activated without further user action. The following system message displays:

```
CCBR020 New Keycode accepted 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. The following system message displays:

```
CCBR009 New Keycode accepted. It is activated during the next
restart.
```

```
CCBR009 New Keycode accepted. It is activated during the next restart.
```

test

If the keycode is not eligible for instant activation, a Sysload is needed to activate the new keycode. The following system message displays:

```
CCBR009 New Keycode accepted. It is activated during the next
restart.
```

CCBR020 New Keycode accepted 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. The following system message displays:

CCBR009 New Keycode accepted. It is activated during the next restart.

5. Access LD 22 and confirm that the new License parameters have been updated.

>LD 22 REQ SLT ....

6. See Reverting to the previous keycode with the KRVR command on page 133 if License limits are not increased or problems exist.

Instant License parameter upgrade for Small Systems

For Small Systems, perform the following steps 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.

   The following questions require information from the Keycode data sheet. Have it available.

5. Indicate that the current Feature Sets and/or Packages 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 License Parameters are printed to the TTY.

6. The License parameters shown below are a sample configuration only.

Current License Parameters : TNS (10) AGNT (10) ACDN (10) AST (10) DSL (10) ...

7. Do you wish to change any License parameters? (y/n/[a]bort) :
8. In response to the prompt "Do you wish to change any License parameters? (y/n/[a]bort) :" enter **y**.
9. The License parameters are prompted in sequence. Change the License parameters appropriately, according to the new keycode:

   The License parameters shown below are a sample configuration only.

   Enter new License parameters, <cr> to leave unchanged: TNS (10) - AGNT (10) - 11 ACDN (10) - AST (10) - DSL (10) - ...
10. After all License parameters have been prompted, the new License parameters are displayed and the prompt "Is this correct?" appears. Enter **y** to continue.
11. New License 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 License parameters are correct. If the License parameters are not correct, select n and reconfigure the License parameters.

    The system displays 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 License limit is shown. In this example the AGNT License limit was changed from 10 -11. The system displays: AGNT: 10 11 ... Is this correct? (y/n/[a]bort):
18. Select **y**. The system prompts for a 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 displays:

    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 is correct for instant activation, it is activated without further user action. The following message displays:

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. 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."

The terms "old" and "new" keycode as discussed here see 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 License 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 is activated without further user action. The following system message displays:

CCBR020 New Keycode accepted 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. The following system message displays:

CCBR009 New Keycode accepted. It is activated during the next restart.

Instant License 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 displays 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 License limits or reduce features compared with the existing keycode. If you have determined that the keycode lowers License limits or reduces features, do not continue with the KNEW command, but contact your Nortel 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 is activated without further user action. The following system message displays:

   CCBR020 New Keycode accepted 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. The following system message displays:

CCBR009 New Keycode accepted. It is 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 133 if License limits are not increased or problems exist.

Chapter 12: Nortel Integrated DECT

Contents

This section contains information on the following topics:

Feature description

Nortel Integrated DECT (DECT) is an application on the system that allows digital wireless capabilities. With 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, see DECT Fundamentals.

Operating parameters

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.

On a CS 1000M Cabinetor Meridian 1 PBX 11C Cabinetsystem, DECT supports a maximum of 630 users. A CS 1000M Chassisor Meridian 1 PBX 11C Chassissupports a maximum of 96 DECT users. Large Systems support a maximum of 1024 users.

Feature interactions

DECT does not require DTI programming in LD 73.

Feature packaging

DECT requires the Meridian Companion Option (MCMO) package 240.

Feature implementation

Task summary list

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

1. Table 13: LD 10 on page 138

   Configure a DECT telephone

2. Table 14: LD 73 on page 139

   Configure DECT pad values

Table 13: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Analog (500/2500-type) telephone.
TN		Terminal number
	l s c u	Format for Large Systemand CS 1000E, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small Systemand Media Gateway 1000B where c = card and u = unit.
...		

Prompt	Response	Description
WRLS	(NO) YES	Indicates that this TN corresponds to a portable personal telephone or Nortel DECT Handset 4060. Only offered if the MCMO package is equipped.
WYTP	(MCMO) DECT	Wireless type assigns the TN to 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 telephones. Only offered if the MCMO package or SUPP package is equipped.
	(DTN)	Default digit signaling used by portable personal telephone.

Table 14: LD 73

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	DECT 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.

Chapter 13: Integrated Messaging System Link

Contents

This section contains information on the following topics:

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 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. Table 15: LD 17 on page 143

   Add or change the link to a messaging system
2. Table 16: LD 17 on page 143

   Add or change the link to a messaging system
3. Table 17: LD 15 on page 144

   Add or change the IMS feature for a customer
4. Table 18: LD 23 on page 145

   Add or change ACD data for Integrated Messaging System Link feature
5. Table 19: LD 11 on page 145

   Add or change IMS attendant capability for each Meridian 1 proprietary telephone

Before adding, changing, or removing a link, the device must be disabled. See Table 15: LD 17 on page 143

Table 15: LD 17

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. If the number of call registers defined for the system (prompt NCR) is within the range 80 to 1020, AXQI and AXQO cannot exceed 25 percent of the system call registers.

Before adding, changing, or removing a link, the device must be disabled. See Table 15: LD 17 on page 143

Table 16: LD 17

Prompt	Response	Description
REQ	CHG	Change

Prompt	Response	Description
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. If the number of call registers defined for the system (prompt NCR) is within the range 80 to 1020, AXQI and AXQO cannot exceed 25 percent of the system call registers.

Table 17: LD 15

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
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.

Prompt	Response	Description
- 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.

Table 18: LD 23

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.

Table 19: LD 11

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
	c u	Format for Small System and Media Gateway where c = card and u = unit.
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 xx MIK xx MCK xx NRD xx MSB	Add an INCALLS key, where: xxxx = IMS Directory Number (DN), and yyyy = Agent ID. IMS DN and Agent ID can have up to seven digits if DN Expansion package is equipped. 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.

Chapter 14: 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, see*ISDN Primary Rate Interface Fundamentals, NN43001-569* and *ISDN Primary Rate Interface Installation and Commissioning, NN43001-301*.

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, see *ISDN Basic Rate Interface Fundamentals, NN43001-380*and *ISDN Basic Rate Interface Installation and Commissioning, NN43001-318*.

Chapter 15: Intelligent Peripheral, Equipment Completion

Contents

This section contains information on the following topics:

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 is 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 is 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 limitiations for trunk-to-trunk connections based on trunk type is 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 are packaged with Intelligent Peripheral Equipment (XPE) package 203.

Feature implementation

Task summary list

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

1. Table 20: LD 16 on page 152

   Configure a RON/TRON signaling trunk route
2. Table 21: LD 14 on page 152

   Configure RON/TRON Signaling trunk

Table 20: LD 16

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

Table 21: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data.
TYPE	TIE	Trunk type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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

Task summary list

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

1. Table 22: LD 16 on page 153

   Configure L1 signaling on a XFEM TIE trunk route
2. Table 23: LD 14 on page 154

   Configure a L1 Signaling trunk

Use Table 22: LD 16 on page 153 to configure L1 signaling on a XFEM TIE trunk route with Proceed to Send expected after an outgoing seize and answer supervision.

Table 22: LD 16

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

Prompt	Response	Description
	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

Table 23: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data.
TYPE	TIE	Trunk type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit
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

Task summary list

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

1. Table 24: LD 16 on page 155

Configure LDR signaling on XIDID trunk route

2. Table 25: LD 14 on page 155

   Configure LDR signaling on XIDID trunk

Table 24: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add new data or change existing data
TYPE	RDB	Route data block
...		
TKTP	TIE	Trunk type
...		

Table 25: LD 14

Prompt	Response	Description
REQ	NEW	New
TYPE	TIE	TIE trunk data block
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit
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.

Chapter 16: Intelligent Peripheral Equipment and Software Support Enhancements

Contents

This section contains information on the following topics:

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. Table 26: LD 14 on page 160

   Configure XFEM BPO trunk type signaling, and the Balance Impedance Adjustment on XDID trunk

2. Table 27: LD 16 on page 160

Configure the Autoguard Repeat Prevention timer for the route

3. Table 28: LD 97 on page 160

   Configure the five XFALC timers to support downloading

Table 26: LD 14

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

Table 27: LD 16

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

Table 28: LD 97

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

Prompt	Response	Description
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

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.

Chapter 17: Intercept Computer Interface

Contents

This section contains information on the following topics:

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 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 telephone. This code specifies the reason for the tenant 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 a digital telephone (deactivation).

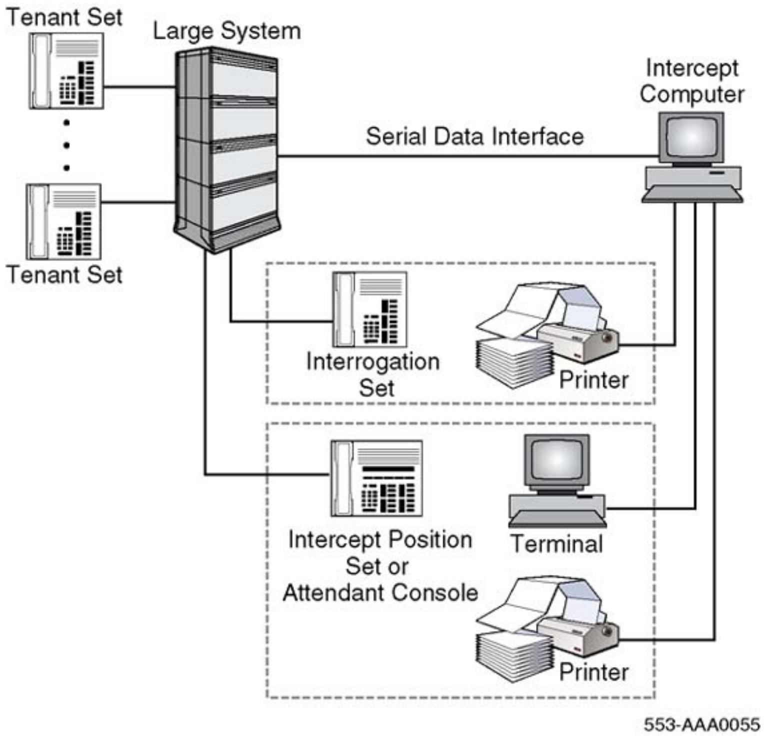


Figure 3: Intercept Computer Interface components

- 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 telephone. 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 telephone may be designated as a channel in the ICP answering machine in LD 10. The telephone must have a Digitone (DTN) Class of Service.

Operating parameters

An analog (500/2500-type) telephone can only be used as an interrogation telephone, 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 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 digital 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 telephone that has the ICP Call Forward feature active is idle, the DN is blocked and if the DN is busy, busy tone is heard.

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

The Intercept Computer Interface (ICP) 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. Table 29: LD 17 on page 166

   Configure the configuration record for ICP
2. Table 30: LD 15 on page 167

   Configure the customer data block for ICP
3. Table 31: LD 10 on page 167

   Create or modify the analog (500/2500-type) telephone data block for ICP
4. Table 32: LD 11 on page 167

   Create or modify the Meridian 1 proprietary telephone data block for ICP
5. Table 33: LD 12 on page 168

   Create or modify the attendant console data block for ICP
6. Table 34: LD 23 on page 168

   Modify the ACD/Message Center parameters for Incoming Call Indicators (ICIs)
7. Table 35: LD 93 on page 168

   Enable or modify the Multi-tenant Service feature

Table 29: LD 17

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration data block
IOTB	(NO) YES	Change to logical units

Prompt	Response	Description
ADAN	NEW TTY x	Add TTY number x

Table 30: LD 15

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.

Table 31: LD 10

Prompt	Response	Description
REQ:	NEW, CHG	Add, or change.
TYPE:	500	500/2500 Telephone data block
...		
CLS	(IRGD) IRGA	Interrogation telephone for Intercept Computer allowed (denied).
	(IAMD) IAMA	Allow a 2500-type telephone to be a channel in the ICP Answering Machine (CLS DTN is required).
ICT	0-NIPN	Terminal/printer number (NIPN configured in LD 15).

Table 32: LD 11

Prompt	Response	Description
REQ:	NEW, CHG	Add, or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS	(IPND) IPNA	Terminal/printer number (NIPN configured in LD 15).

Prompt	Response	Description
ICT	0-NIPN	Terminal/printer number (NIPN configured in LD 15).

Table 33: LD 12

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).

Table 34: LD 23

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ACD	ACD data block
...		
ICP	(NO) YES	ACD MC (is not) is an intercept position.
ICPS	COM (CIR)	Intercept Computer printer search Common printer for ACD group Circular hunt

Table 35: LD 93

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) COM	Circular search. 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 message for the tenant DN and activate the message waiting LED at the tenant 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).

Chapter 18: Intercept Computer Dial from Directory

Contents

This section contains information on the following topics:

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 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 (because 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 telephone 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 telephone 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 (because 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 telephone.

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 telephone 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 because 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 telephone 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. Table 36: LD 15 on page 177

   Allow or deny an intercept attendant to dial an extension DN from the ICP
2. Table 37: LD 15 on page 178

   Set the switchhook flash time required when using package 131
3. Table 38: LD 21 on page 178

   - Print Intercept Computer Dial from directory system information

Table 36: LD 15

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".

Table 37: LD 15

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	TIM	Timers data block.
...		
- FLSH	xxx yyy	Switch Hook Flash timer.

Table 38: LD 21

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. Telephone 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. Telephone 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. Telephone 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). Telephone 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. Telephone 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).

Chapter 19: Intercept Computer Enhancements

Contents

This section contains information on the following topics:

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 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 telephone 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:

- Table 39: LD 10 on page 185

  Configure Intercept Computer Answering Machine Class of Service
- Table 40: LD 15 on page 185

  Configure internal and external call DNs for Intercept Transfer
- Table 41: LD 93 on page 185

  Configure internal and external call DNs for attendant console groups

Table 39: LD 10

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	500/2500 telephone data block.
...		
CLS	(IAMD) IAMA	ICP Answering Machine (denied) allowed. Allow a 2500 telephone to be a channel in the ICP Answering Machine.

Table 40: LD 15

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.

Table 41: LD 93

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.

Prompt	Response	Description
		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.

Chapter 20: Intercept Treatment

Contents

This section contains information on the following topics:

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 42: Intercept Treatment for various types of calls on page 188.

Table 42: 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 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 43: Intercept Treatment for BARS/NARS calls on page 189 specifies the type of Intercept Treatments (INTR) available for BARS/NARS calls, and lists the intercept situations that are possible.

Table 43: Intercept Treatment for BARS/NARS calls

Intercept situation	Station or DISA	Originating party		CCSA/DID trunk
		Attendant extended calls	TIE trunk (including attendant)	
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) 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

Table 44: LD 15 - Change customer Intercept Treatment for various call types

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	INT	Intercept treatment options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
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		RAN route number for "Authcode Last" prompt (NAUT)

Prompt	Response	Description
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 21: Intercept Treatment Enhancements

Contents

This section contains information on the following topics:

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

Table 45: 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.

Chapter 22: International Meridian 1

Contents

This section contains information on the following topics:

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 are 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. Table 46: LD 17 on page 200

   Configure the configuration record for 16-Button Dual-tone Multifrequency (DTMF) detection
2. Table 47: LD 13 on page 200

   Configure DTR, TDET, and DTD cards for this feature
3. Table 48: LD 97 on page 200

Configure system parameters for Intelligent Peripheral Equipment in configuration record 2

Table 46: LD 17

Prompt	Response	Description
REQ	CHG	Request.
TYPE	PARM	Change system parameters.
...		
-ABCD	(NO) YES	16-tone DTMF operation enabled.

Table 47: LD 13

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		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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		To Terminal Number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.

Table 48: LD 97

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SYSP	System parameters.
...		

Prompt	Response	Description
INTN	(NO) YES	?-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.

Chapter 23: Inventory Reporting

Contents

This section contains information on the following topics:

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.

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 49: Inventory Reporting generate commands on page 203 for a list of commands and their descriptions.

Table 49: Inventory Reporting generate commands

Command	Description
INV GENERATE ABORT	Abort all Inventory generations.

Command	Description
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 51: Card inventory example for Meridian 1 Option 61C on page 205 or Table 52: Telephone inventory example for Small Systems on page 205 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 50: ID PROM information on page 204 for more information.

Table 50: 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

The Small System card TN only includes the card number.

See Table 51: Card inventory example for Meridian 1 Option 61C on page 205 for an example.

Table 51: 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
....

<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 TN (loop, shelf, card, and unit numbers)
- 32 bytes of ID PROM information
- the device descriptor information (DES field in LDs 10 and 11)
- the primary DN

The Small Systemtelephone TN only includes a card number and a unit number

See Table 52: Telephone inventory example for Small Systems on page 205 for an example.

Table 52: Telephone inventory example for Small 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
....          ....          ....                                  ....          ....
```

<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 53: Inventory Reporting midnight routine commands on page 206 for a list of commands and their descriptions.

Table 53: 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 54: Inventory Reporting print commands on page 207 for a list of commands and their descriptions.

Table 54: 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

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 55: Inventory Reporting status responses on page 208 for a list of status responses and their descriptions.

Table 55: 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 56: 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
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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
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
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	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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
FXNET	Fiber Extended Network	NTIP61	BA	Global
FXPEC	Fiber Extended Peripheral Equipment	NTIP62	CA	Global
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
RAN	Integrated Recorded Announcer	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
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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
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
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,	Global

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
			LB, LC, LD, MA, MB, MC, NB, NC, PA, PB, PC, QA, QB, QC, SA, SB, TA, TB	
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
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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
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
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 (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:

- 500/2500 telephones
- Other digital telephones or any non-Nortel (third-party) telephones, including those designed to simulate included telephones.

Data units

A data unit 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
- 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.

Chapter 24: IP Ad Hoc Conference

Contents

This section contains information on the following topics:

Feature description

IP Ad Hoc Conference emulates the same features and functionality as TDM-based Conference by adding additional parties to an established call. It provides 30 slots per IP conference loop.

Operating parameters

The IP Media Services Ad Hoc Conference application is installed with the IP Media Services Controller application.

On the Call Server, the IP-based loop type (IPCONF) emulates conference functionality for extended Virtual TNs.

The IP Media Sessions license reflects the overall number of possible IP sessions with MAS. It is the sum of IP Music + IP Announcement + IP Ad Hoc Conference + IP Attendant Console + IP Tone.

IP Ad Hoc Conference licenses are controlled by the number of available IP Media Services licenses. The number of licenses does not decrement until an IP Ad Hoc Conference loop is used. Each conference participant requires 1 license.

Before you can use this feature, you must configure the Media Application Server (MAS) and Network Routing Service (NRS) for IP Media Services. For information about configuring MAS and NRS for IP Media Services, see *Signaling Server IP Line Applications Fundamentals (NN43001–125)*.

Conference loop selection

Traditional TDM and IP conference loops are grouped together in a common pool of conference resources. The Call Server selects the conference loop based on the caller who presses the Conference function key to initiate the conference. For example, if a caller from an IP-endpoint initiates the conference, an IP-based conference loop (IPCONF) is preferred. Likewise, if a caller from a TDM endpoint completes the conference loop, a TDM conference loop (CONF) is preferred.

Although the system attempts to match the loop type with the service target, the loop selection for both IP and TDM endpoints is based on resource availability. For example, if there are no IP conference loops available for an IP-based endpoint, the system selects an available TDM conference loop. This also applies to TDM endpoints. If there are no TDM conference loops available for a TDM-based endpoint, the system selects an available IP conference loop. Loops are enabled by default. Disabled loops or unregistered IP conference loops are not considered for selection.

Feature interactions

The feature interactions for IP Ad Hoc Conference are the same as those for TDM Conference. For information about Conference feature interactions, see *Features and Services, Book 2, NN43001-106*.

Feature packaging

The IP Ad Hoc Conference application is included with IP Media Services Package 422.

Feature implementation

You can add, modify, remove, or check the status of IP conference loops using the overlay commands described in the following section.

Note:

IP Ad Hoc Conference operations (configuration, removal, printing, and so on) are only available for unrestricted IP Media Services packages.

Overlays

Task summary list

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

- Table 57: LD 17 - Create or modify an IP Ad Hoc Conference loop on page 219
- Table 58: LD 38 - Enable or disable IP Ad Hoc Conference loop on page 220
- Table 59: LD 117 - Print status of IP Ad Hoc Conference application on page 220

Table 57: LD 17 - Create or modify an IP Ad Hoc Conference loop

Prompt	Response	Description
REQ	xxx	NEW or CHG
TYPE	CEQU	Common Equipment parameters
...		
IPCONF	0-255	Virtual IP conference loop number The maximum number of loops is 64 (this total includes CONF, TONE, IPCONF, IPTONE, or any combination thereof). You can add multiple loops at the same time. Precede a loop number with X to remove it. You can remove multiple loops at the same time. **Note:** You must first disable a loop before removing it.
NODE	1-9999	Node ID of the IP Ad Hoc Conference loop.

Table 58: LD 38 - Enable or disable IP Ad Hoc Conference loop

Command	Description
ENLL <loop>	Enable the specified conference loop
DISL <loop>	Disable the specified conference loop
STAT <loop>	Print the status of the specified conference loop Output format for IP loop is: • IPCNFC n DSBL n BUSY n REG = number of IP conference groups disabled and busy and the registration status of the IP Media Services Conference Controller, where: - 00 = IP Media Services Conference Controller is not properly registered with the Call Server - 16 = Maximum number of IP conferences are registered and available for use • IPCHAN n DSBL n BUSY n REG = number of IP channels disabled and busy and the registration status of the IP Media Services Conference Controller, where: - 00 = IP Media Services Conference Controller is not properly registered with the Call Server - 30 = Maximum number of IP channels are registered and available for use

Table 59: LD 117 - Print status of IP Ad Hoc Conference application

Command	Description
STAT SERV	Prints the status of all servers registered to the Call Server.
STAT SERV APP IPCONF	Prints only the status of servers that have IP Ad Hoc Conference listed as an application.

Element Manager

You can add, delete, and check the status of IP Ad Hoc Conference loops using the Element Manager Loops page.

Adding an IP Ad Hoc Conference loop using Element Manager

1. To configure or edit Loops information, click the **Core Equipment > Loops** link of the System branch of the Element Manager navigator.

   The Loops Web page appears.

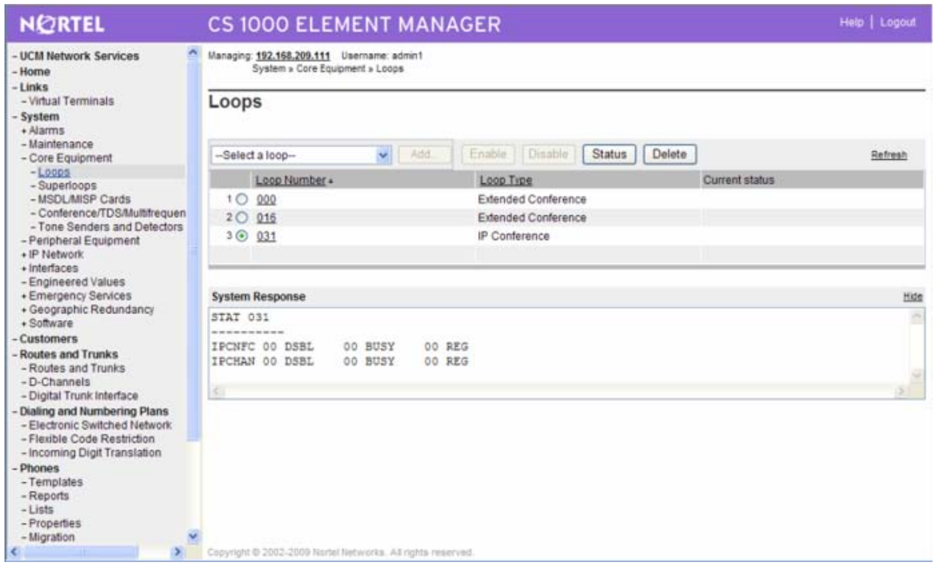


Figure 4: Element Manager Loops page

2. From the Select a loop list menu, select IP Conference.

   The IP Conference Loop Number Details page appears.

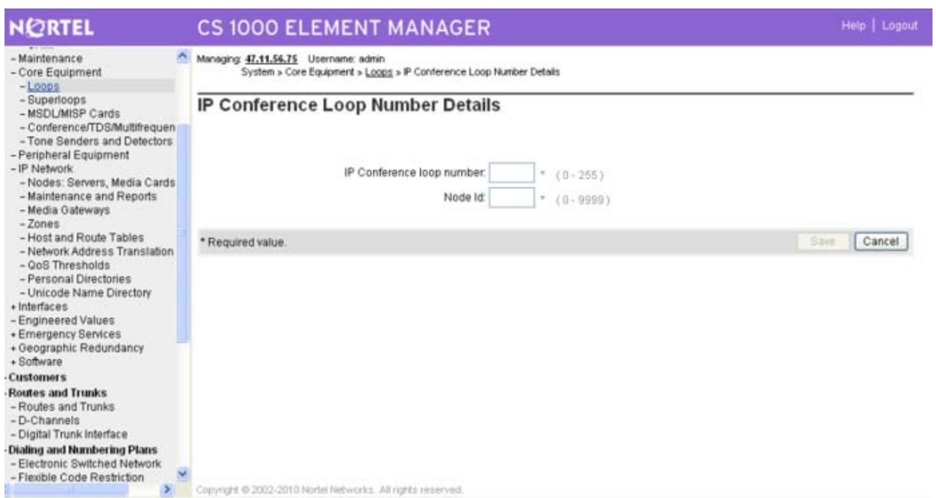


Figure 5: IP Conference Loop Number Details page

3. Enter the IP Conference loop number. The value can be 0-255.
4. Enter the Node ID. The value can be 0-9999.
5. Click **Save**.

Feature operation

No specific operating procedures are required to use this feature over and above the operating procedure for the existing base conference feature.

Chapter 25: IP Attendant Console

Contents

This section contains information on the following topics:

Feature description

The IP Attendant Console 3260 is an IP-enabled Attendant Console that replaces the need for a Personal Computer Console Interface Unit (PCCIU) or M2250 Digital Attendant Console for supported third party Attendant Console clients. An Avaya provided Software Development Kit (SDK) must be used to enable IP signaling and voice for third-party Attendant clients. The IP Attendant Console is included with the IP Media Services applications that are installed as part of the Signaling Server software.

Note:

The IP Attendant Console 3260 is not supported for Communication Server 1000M systems.

The IP Attendant Gateway uses Session Initialization Protocol (SIP) to manage signaling between the IP Attendant Console and the Media Application Server (MAS). Communication with the Call Server is managed using the existing PC TCM messaging, which is now over TCP.

The IP Attendant Console 3260 is configured on a virtual loop. You can configure up to a total of 63 attendant consoles (of type M2250, IP 3260, or a combination of both) on each system. You can only configure the number of consoles permitted by ISM License parameters.

The IP Attendant Console 3260 has the following predefined features:

Table 60: Predefined features of the Attendant Console 3260

Mnemonic	Label
C/H	Centralized Auto-Attendant Service
B/N	Position Busy/Night Service
excl src	Exclude Source
excl dest	Exclude Destination
conf 6/BLF	Conference 6/Busy Lamp Field
RLS src	Release source
RLS dest	Release destination
SHIFT	Shift key
HOLD	Hold
lpk 0	Loop pickup 0
lpk 1	Loop pickup 1
lpk 2	Loop pickup 2
lpk 3	Loop pickup 3
lpk 4	Loop pickup 4
lpk 5	Loop pickup 5
RLS	Release

Supported features and options

The Attendant Console 3260 supports the following KEY (LD 12) features:

Table 61: Supported KEY (LD12) features for the IP Attendant Console 3260

Mnemonic	Description
ADL	Autodial
AWU	Automatic Wake Up
BIN	Barge-In
BKI	Break-In
BVR	Busy Verify
CHG	Charge Account
COS	Controlled Class of Service

Mnemonic	Description
CPN	Calling Party Number
DCW	Display Call Waiting
DDL	Do Not Disturb Individual
DPD	Display Destination
DPS	Display Source
DRC	DID Route Control
EES	End-to-End Signaling
GND	Group Do Not Disturb
MCK	Message Cancellation
MDT	Maintain Date/Display Date
MIK	Message Indication
MTM	Maintain/Display Time
MTR	Meter
NAS	Network Attendant Service
PRK	Call Park
RDL	Redial stored number
RFW	Attendant Remote Call Forward
RPAG	Radio Paging
RTC	Routing Controls
SCC	Speed Call Controller
SECL	Series Call
SSC	System Speed Call Controller
TRC	Malicious Call Trace

For additional supported key features, see the entry for Attendant Consoles in *Features and Services Fundamentals—Book 1 of 6, NN43001-106*.

Non-supported features and options

The Attendant Console 3260 delivers most of the features and functionality of a M2250 attendant console. However, the Attendant Console 3260 does not support the following features and options:

Non-supported KEY (LD 12) features:

- AUTO—Direct Autoline DN
- SACP—Semi-Automatic Camp-On
- THF—Trunk Switch Hook Flash

- ABDN—Activation of Attendant Blocking DN
- AFBT—Attendant Forward Buzz Tone
- ATAC—Attendant Administration Access Code
- EFLL—Efficiency Factor Loading Level
- MATT—Consoles used as Message Center
- RICI—ICI Keys to receive Recorded Overflow Announcement
- SPVC—Supervisory Console

Non-supported OPT (LD 15) options:

- ABDA (ABDD)—Attendant Busy Display Allowed/Denied
- AMA (AMD)—Attendant Monitoring Allowed/Denied
- IBL (XBL)—Include or Exclude Enhanced busy Lamp Field
- ILF (XLF)—Include or Exclude Lamp Field Array
- EBLF—Enhanced Busy Lamp Field
- PSA (PSD)—Presentation Status Selection on Attendant Console Allowed/Denied
- SLA (SLD)—Slow Answer Recall Enhancement Allowed/Denied

For attendant console features and options, see *Features and Services Fundamentals—Book 1 of 6, NN43001-106*.

Network management

Networking issues, such as packet loss, delay, and jitter, may sometimes affect the quality of service for IP-based voice calls. The following table lists common networking issues and thresholds for IP Attendant Console 3260 voice quality:

Table 62: Networking issues and thresholds

Item	Description
Packet Loss	When a call is established using a G.711 codec and there is a network packet loss of 0.5%, there is no impact to voice quality.
IP Packet Delay	When a call is established using a G.711 codec and there are end-to-end delays of 150 milliseconds (network delay, packet delay, and jitter buffer delay), there is no impact to voice quality.

Item	Description
Jitter	When a call is established using a G.711 codec and there is a jitter buffer of 100 milliseconds, there is no impact to voice quality.

Note:

QoS and DiffServ voice packet prioritization are not supported. Due to these limitations, you must ensure the network is engineered to guarantee low packet loss and latency.

Deployment options

In a Direct deployment, the IP Attendant client communicates directly with an IP Attendant Gateway.

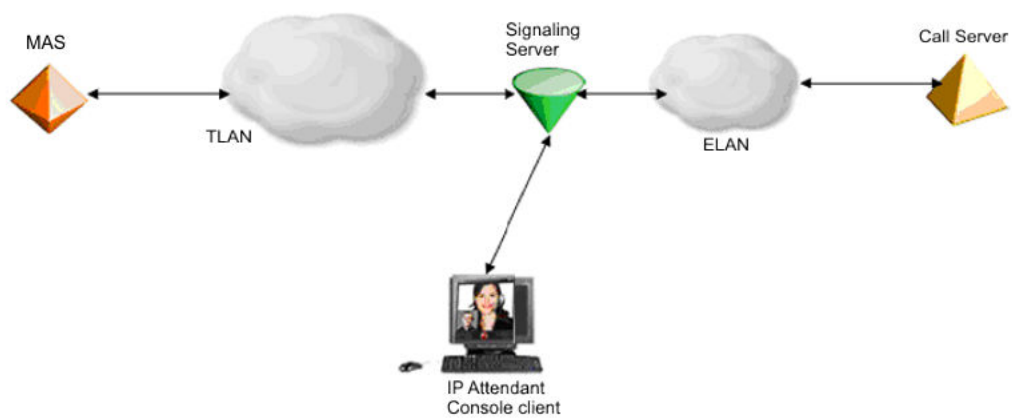


Figure 6: Direct deployment using 1 Call Server and 1 Signaling Server

In a Redirect deployment, the IP Attendant Console 3260 client uses NRS to determine the appropriate IP Attendant Gateway to communicate with. If the system uses a Network Routing Service (NRS), you must create a non-dialable entry to specify the IP address and port of each IP Attendant console in the network. If the network has multiple Call Servers, the TN for each IP Attendant Console 3260 must be identical on each Call Server they have access to.

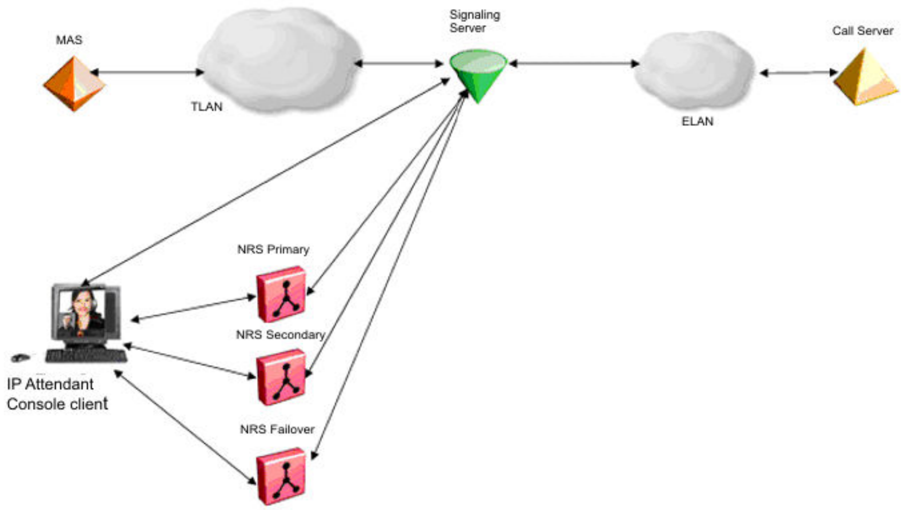


Figure 7: Redirect deployment using 1 NRS, 1 Signaling Server, and 1 Call Server

Planning considerations

When planning to configure IP Attendant, the IP Media Services feature must be enabled and configured. In addition, each IP Attendant Console 3260 requires 3 media services sessions on the MAS. Also, the network must be engineered to support 4 additional SIP signalling sessions. You must ensure that there are enough media services sessions and SIP ports to support the intended number of consoles.

IP Attendant requires the use of a third-party client. For information about configuring a third-party client to interface with the IP Attendant Console 3260, refer to the third-party client documentation. The ports entered for the third-party client software must match the ports entered for IP Attendant in the Element Manager IP Media Services configuration page.

For information about CS 1000 planning considerations, including traffic estimation calculations, see *Communication Server 1000E Planning and Engineering, NN43041-220*.

Licensing requirements for IP Attendant Console

The minimum licensing requirements for the IP Attendant Console are as follows:

- Third party IP Attendant client license
- 1 IP Attendant 3260 license for each IP Attendant console
- 4 SIP Ports for the IP Attendant Gateway
- Package 422 must be enabled on the Call Server in the keycodes

- 1 MAS Nodal license for each MAS server in the network
- 3 IP Media Session licenses for each IP Attendant Console

The IP Media Sessions license reflects the overall number of possible IP sessions with MAS. It is the sum of IP Music + IP Announcement + IP Ad Hoc Conference + IP Attendant Console + IP Tone.

For an IP Attendant Console to function, you require one IP Attendant Console license and three IP Media Services sessions. For each installed console, the total number of available IP Media Sessions licenses decrements by three.

IP Attendant Consoles

Each IP Attendant Console 3260 in the system requires 4 SIP ports. You must ensure that there are enough SIP ports to support the intended number of consoles.

Traffic estimation

You only need to perform traffic estimation calculations if the overall bandwidth between the IP Attendant Consoles and the registered IP Media Services server is less than 20 Mbps. This applies to all deployments.

The traffic estimation calculations shown in the table below are only applicable to the traffic between the IP Attendant Console 3260 and the IP Media Services server.

Table 63: Traffic estimation calculations for IP Attendant Console 3260

Attendants	Estimated Traffic Load (Mbps)	Load Requirements
1	0.128	0.13
2	0.256	0.26
3	0.384	0.38
4	0.512	0.51
5	0.64	0.64
10	1.28	1.3
15	1.92	1.92
20	2.56	2.6
30	3.84	3.8
40	5.12	5.1

Attendants	Estimated Traffic Load (Mbps)	Load Requirements
50	6.4	6.4
60	7.68	7.7
63	8.06	8.1

Operating parameters

The IP Attendant Console 3260 requires that a MAS be part of the CS 1000 network. For information about MAS, see *Media Application Server Platform Fundamentals, NN44471-101*.

Before you can use this feature, you must configure the Media Application Server (MAS) and Network Routing Service (NRS) for IP Media Services. For information about configuring MAS and NRS for IP Media Services, see *Signaling Server and IP Line Applications Fundamentals (NN43001–125)*.

Feature interactions

The IP Attendant Console 3260 does not support Secure Real-time Transport Protocol (sRTP) or Transport Layer Security (TLS) signaling encryption.

Feature packaging

The IP Attendant Console 3260 is packaged with the IP Media Services applications and installed as part of the Signaling Server application.

Feature implementation

You configure the IP Attendant Console 3260 the same way as the 2250 Console, with the exception of the following prompts:

- IPCR—the IP Call Recording feature is not implemented for the IP Attendant Console 3260.
- Zone—you must configure this parameter. The IP Attendant Console 3260 uses the same Zone, including SRN, as the rest of IP Media Services.

The following list describes the steps and requirements for configuring an IP Attendant Console 3260.

- IP Media Services package 422 must be enabled. In addition, you must meet the ISM licensing requirements.
- IP Attendant must be configured on the Signaling Server as part of IP Media Services. For information about configuring IP Attendant as part of IP Media Services, see *Signaling Server IP Line Applications Fundamentals, NN43001-125*.
- IP Attendant must be configured on the Media Application Server (MAS) as part of IP Media Services. For information about configuring MAS applications, see *Application Management, NN44471-601*.
- Configure the IP Attendant 3260 client with the required parameters. For information about configuring the client, refer to the documentation that shipped with your product.

Note:

Third-party clients may or may not support all IP Attendant Console 3260 features. For information about client supported features, refer to the client documentation.

Task summary list

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

- Table 64: LD 12 - Add or modify an IP Attendant 3260 console on page 232
- Table 65: LD 20 - Print IP Attendant Console 3260 details on page 232
- Table 66: LD 22 - Print the total number of used and available IP Attendant Consoles on page 233
- Table 67: LD 81 - Print the number of IP Attendant 3260 consoles on page 233
- Table 68: LD 117 - Print IP Attendant information on page 233

Table 64: LD 12 - Add or modify an IP Attendant 3260 console

Prompt	Response	Description
REQ	aaa	NEW, CHG, MOV, or OUT
TYPE	3260	IP Attendant Console 3260
TN	l s c u	Terminal number
SETN	l s c u	Second TN number
ANUM	1-63	Attendant number **Note:** You cannot assign 0 to an attendant.
SSU	0-4095	System Speed Call user list number
ICDR	(ICDD) ICDA	(Deny) allow internal call detail
CPDN	(CNDD) CNDA	(Deny) allow Call Party Name Display
DNDI	(DNDD) DNDA	(Deny) allow dialed name display
...		
AADN	xxx...x	Attendant Alternative Answering DN
ZONE	<number>	Zone bandwidth number
...		
KEY	xx aaa yyyy	Key definition, where: • xx = key number • aaa = key name or function • yyyy = additional information required for the key

For a list of Attendant Console key functions, see the entry for Attendant Console in *Features and Services Fundamentals, Book 1, NN43001-106*.

Table 65: LD 20 - Print IP Attendant Console 3260 details

Prompt	Response	Description
REQ	PRT	Print
TYPE	3260	IP Attendant Console 3260
TN	l s c u	Terminal number

Table 66: LD 22 - Print the total number of used and available IP Attendant Consoles

Prompt	Response	Description
REQ	SLT	Print System Type, System Generic, and System Limits. For CS 1000E systems, displays a summary status of all IPMGs configured on the system. Output includes the total of used and available IP Attendant Consoles.

Table 67: LD 81 - Print the number of IP Attendant 3260 consoles

Prompt	Response	Description
REQ	CNT	Count
CUST	nn	Customer number, as defined in LD 15
...		
FEAT	3260	Attendant Console 3260

Table 68: LD 117 - Print IP Attendant information

Command	Description
PRT IPDN <IP_address>	Print IP Attendant Console 3260 information for the specified IP address
STAT IP TYPE 3260	Print IP Attendant Console 3260 status
STAT SERV	Display status of all servers
STAT SERV APP IPATTN	Display status of servers hosting the IP Attendant Console 3260

Chapter 26: IP Music

Contents

This section contains information on the following topics:

Feature description

IP Music provides the same functionality and features as TDM Music Broadcast. The IP Media Services music service can provide music for all CS 1000 features that require a music source as it uses the traditional music route and trunk interface that the Call Server uses to obtain music.

The IP Music broadcast feature allows multiple calling parties to receive IP Music from one IP Music trunk. Each registered IP Music trunk can provide a maximum of 60 broadcast connections.

Operating parameters

On the Call Server, the IP Music route type (IMUS) provides music to callers using existing virtual TNs on analog trunks. The IP Music trunk route type contains virtual TN-based analog trunks as its trunk members.

The IP Media Sessions license reflects the overall number of possible IP sessions with MAS. It is the sum of IP Music + IP Announcement + IP Ad Hoc Conference + IP Attendant Console

+ IP Tone. The IP MUS license works like the traditional digital MUS license. Available licenses are not decremented until a MUS trunk is used, at which time they are decremented by 1 up to a maximum of 60 licenses for each Trunk member. IP Media Services licenses are decremented simultaneously.

Before you can use this feature, you must configure the Media Application Server (MAS) and Network Routing Service (NRS) for IP Media Services. For information about configuring MAS and NRS for IP Media Services, see *Signaling Server and IP Line Applications Fundamentals (NN43001–125)*.

Feature interactions

IP Music has the same feature interactions as the TDM Music Broadcast feature. For more information about Music, see the entry for Music in *Features and Services, Book 4, NN43001-106*.

Feature packaging

The IP Music application is included with IP Media Services Package 422.

Feature implementation

Trunk units for IP Music route data blocks are configured on virtual superloops using the same overlays used to configure traditional music trunks. The density of the IP Music card is extended to 32 units, so you can configure units 0 to 31.

Note:

If you are configuring IP Music and IP RAN trunks on the same card number in the system, you must keep the two types separated from each other in blocks of 4 units. For example, if you configure IP Music trunks in units 0-3, the system does not permit the configuration of IP RAN trunks in the same group of 4 units.

Overlays

Task summary list

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

Table 69: LD 16 - Create or change an IP Music route data block

Prompt	Response	Description
REQ	aaa	NEW or CHG
TYPE	RDB	Route Data Block
CUST	xx	Customer number associated with the route
ROUT	x...x	Route number
...		
TKTP	IMUS	IP Music route data block
ZONE	0-255	Zone for codec selection and bandwidth management
NODE	xxxx	Node ID
ICOG	OGT	Outgoing trunk only
BDCT	YES (NO)	Enable broadcast capability for this route. **Note:** For CS 1000E systems, the default is YES. CS 1000E systems only support broadcast trunks.

Table 70: LD 16 - Remove an IP Music route data block

Prompt	Response	Description
REQ	OUT	Remove data block
TYPE	RDB	Route Data Block
CUST	xx	Customer number associated with the route
ROUT	x...x	Route number

Table 71: LD 14 - Configure a trunk for an IP Music route data block

Prompt	Response	Description
REQ	NEW	New
TYPE	IMUS	IP Music trunk
TN	l s c u	Terminal number
DES	IPMUS	Designator field (optional)
XTRK	IPMS	Extended trunk
CUST	xx	Customer number, as defined in LD 15
RTMB	xxx xxxx	Route number and Member Number

Table 72: LD 11 - Assign MRT to a digital telephone

Prompt	Response	Description
REQ	aaa	NEW or CHG
TYPE	bbb	Type of terminal, where bbb is any BCS or PBX phone.
CUST	xx	Customer number, as defined in LD 15.
...		
MRT	n	Music route number. The route type can be MUS or IMUS. The default is none.

Table 73: LD 11 - Enable Music on Hold for a digital telephone

Prompt	Response	Description
REQ	aaa	NEW or CHG
TYPE	bbb	Type of terminal, where bbb is any BCS or PBX phone.
CUST	xx	Customer number, as defined in LD 15.
...		

Prompt	Response	Description
CLS	(SBMA)/SBMD	Class of service, where: • SBMA = Set Based Music Allowed • SBMD = Set Based Music Denied
...		

Table 74: LD 117 - Check status of IP Music application

Command	Description
STAT SERV APP IPMUS	Prints the status of servers that have IP Music listed as an application.

Element Manager

IP Music trunks and routes can be configured using the **Routes and Trunks** branch of the Element Manager navigator.

Configuring an IP Music route and trunk using Element Manager

1. In the Element Manager navigation tree, click the **Routes and Trunks** link.
2. Click **Add route**.

   The New Route Configuration page appears.

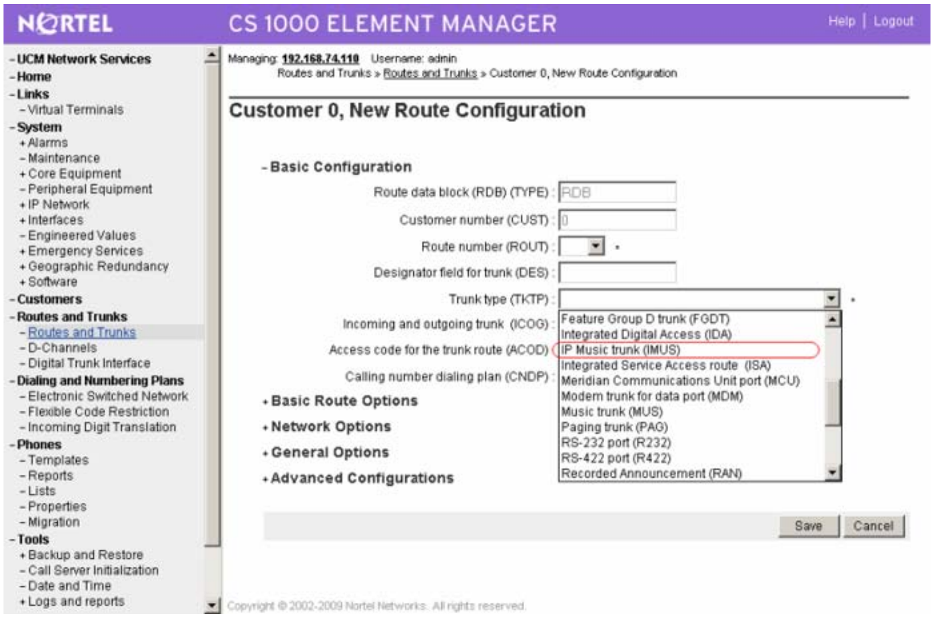


Figure 8: New Route Configuration page

3. In the Basic Configuration section, select **IP Music Trunk (IMUS)** for the Trunk type.

   The virtual trunk route (VTRK) option is automatically selected when creating an IP Music route, as shown in the following figure.

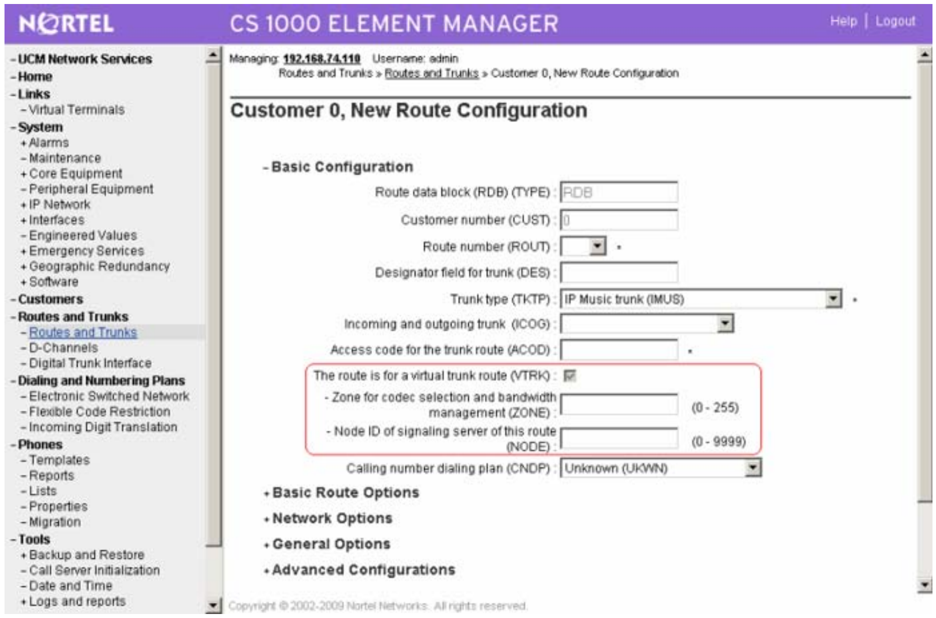


Figure 9: VTRK selected for IP Music route trunk type

4. Configure the route properties and click **Save**.

   The Property Configuration page appears, as shown in the following figure.

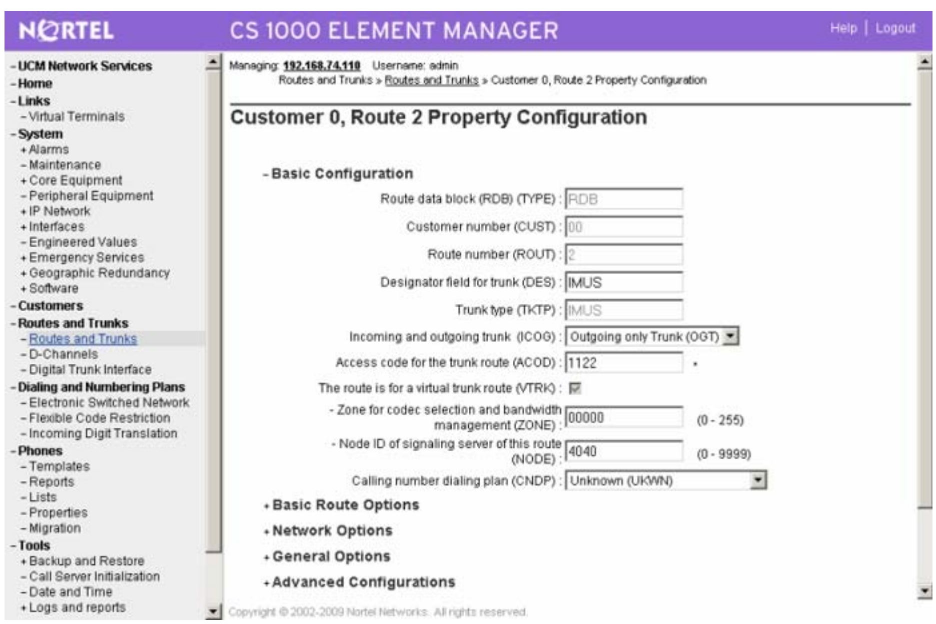


Figure 10: Property Configuration page

5. Configure the route properties and click **Save**.

   The New Trunk Configuration page appears, as shown in the following figure.

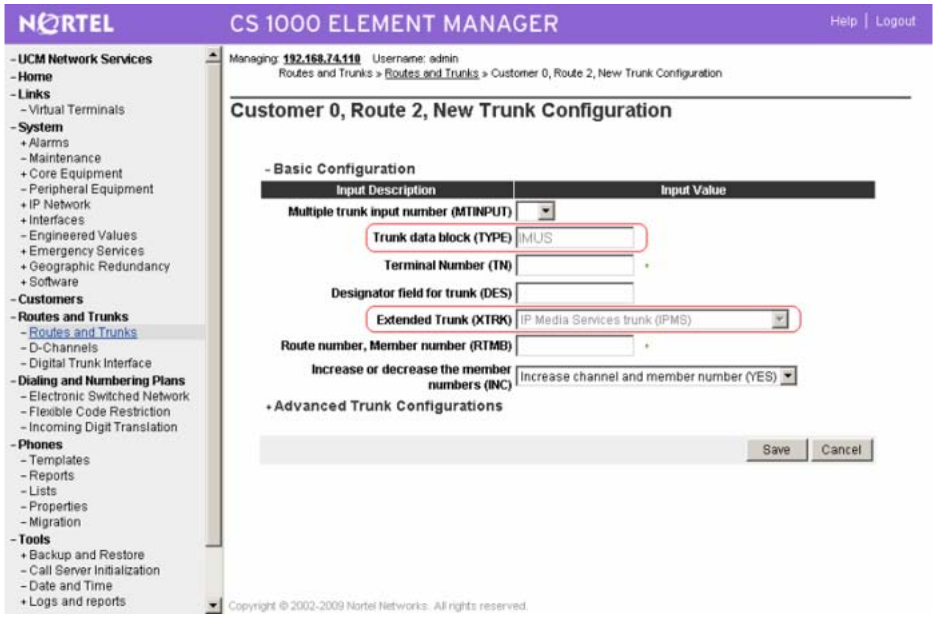


Figure 11: New Trunk Configuration page

6. The Trunk data block type is IMUS and the Extended Trunk (XTRK) is configured as IP Media Services trunk (IPMS). These values cannot be changed.
7. Click **Save**.

For more information about Element Manager, see *Element Manager System Reference - Administration, NN43001-632*.

Feature operation

No specific operating procedures are required to use this feature over and above the operating procedures for the existing base music broadcast feature.

Chapter 27: IP Network-wide Virtual Office

Contents

This section contains information on the following topics:

Feature description

The IP Network-wide Virtual Office feature enables users to log into any IP Phone using their own User ID and password. This redirects the end user telephone calls and other features to the Virtual Office logged-in IP Phone. The end users can perform most IP Phone functions exactly as if they were using their regular IP Phone.

For information on the Virtual Office feature for M3900 series telephones, see M3900 (Single Site) Virtual Office on page 375. For more information on Phantom TNs and Virtual TNs, see Incremental Software Management on page 764.

Upon logon, the IP Phone unregisters with the Call Server and registers to the Call Server associated with the given User ID. This can be the same Call Server or another Call Server within the network. If it is another Call Server, a Gatekeeper is required to provide the address of the IP Phone Terminal Proxy Server (TPS) node. Upon logout from IP Network-wide Virtual Office, the IP Phone registers to the Call Server associated with the settings stored in its EEPROM: the S1 server and Terminal Number (TN).

In order to logon using IP Network-wide Virtual Office, the TN associated with the current IP Phone 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, see LD 11 and LD 81 in *Software Input Output - Administration, NN43001-611*.

Virtual Office is supported for the IP Phone 2001, IP Phone 2002, IP Phone 2004, IP Phone 2007, IP Audio Conference Phone 2033, IP Phone 1110, IP Phone 1120E, IP Phone 1140E, IP Phone 1150E, IP Softphone 2050, MVC 2050, and WLAN Handset 2210/2211/2212. The IP Phone 2002, IP Phone 2004, IP Softphone 2050, and Mobile Voice Client (MVC) 2050, IP Phone support the IP Network-wide Virtual Office feature. For more information, see *Signaling Server IP Line Applications Fundamentals, NN43001-125* or the applicable User Guide.

The IP Network-wide Virtual Office feature supports only one customer group. If more than one customer group is configured, the customer group with the lowest customer number is supported.

The Emergency Services for Virtual Office feature allows IP Network-wide Virtual Office users to place an emergency (E911) call to the correct Public Safety Answering Point (PSAP) for their geographic location. For more information on the Emergency Services for Virtual Office feature, see *Emergency Services Access Fundamentals, NN43001-613*.

When an IP Phone performs an IP Network-wide Virtual Office login to another IP Phone on the same Call Server, but is registered with a different Signaling Server, or Media card, the preempted telephone is logged out with the Logged Out screen. After the login telephone has logged out, the preempted telephone is returned to the home TN in its prior login state.

Operating parameters

The following is the minimum software and hardware required for IP Network-wide Virtual Office:

- Succession 3.0 software and hardware
- IP Line 3.1
- IP Softphone 2050 software build 338 or later
- Element Manager

Firmware upgrade

The IP Network-wide 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 an IP Network-wide Virtual Office login.

The umsUpgradeAll command has no impact on IP Phones logged into IP Network-wide Virtual Office. These telephones do not get reset. If the IP Network-wide Virtual Office login is within the same Call Server, the IP Phone firmware is upgraded after the user has logged out. If the IP Network-wide Virtual Office login is between different Call Servers, the IP Phone simply registers back to its home TPS, and follows the normal firmware upgrade rules for regular registration.

Feature interactions

Automatic Call Distribution

Automatic Call Distribution (ACD) agents requiring the IP Network-wide Virtual Office login, must have a separate, single appearance DN assigned to a programmable feature key. This unique DN is the User ID for IP Network-wide Virtual Office login.

Branch Office DN

When an IP Network-wide Virtual Office login is for a Branch Office DN, the branch office IP Phone is preempted. When the branch office IP Phone is in survival mode and there is an IP Network-wide Virtual Office login for that DN, the branch office DN becomes registered to both the main office and the branch office. This situation is resolved when WAN connectivity resumes and the IP Phone, in local mode, is redirected to register with the Main Office TPS in a logged out state.

Branch Office using Unified Dialing Plan

An IP Network-wide Virtual Office login using Unified Dialing Plan (UDP) at a Branch Office is supported; however, it will fail if the WAN connection to the Main Office is down.

Gatekeeper end points

IP Network-wide Virtual Office support is limited to IP Phones registered with Call Servers that are configured as end points in the same Gatekeeper. IP Network-wide Virtual Office login between IP Phones registered with Call Servers that are configured as end points of different Gatekeepers is not supported.

IP Phone options

The IP Phone options are retained within the telephone regardless of the registered User ID. Remote IP Network-wide Virtual Office logins display the time and date of the remote Call Server.

A Virtual Office user cannot use IP Network-wide Virtual Office to log on to a TN on a Call Server that does not support the IP Phone type of the Virtual Office user For example, CS 1000 Release 5.0 does not support the IP Phone 1200 Series, therefore an IP Phone 1200 Series Virtual Office user cannot perform an IP Network-wide Virtual Office log on to a TN that is located on a CS 1000 Release 5.0.

Multiple Appearance Directory Number

Upon IP Network-wide Virtual Office login using a Multiple Appearance Directory Number (MADN), the TN is selected to match the provided Station Control Password (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).

Phantom Terminal Numbers

See Incremental Software Management on page 764 for more information.

Registration lockout

If a user enters an incorrect password three times, the TN is locked from IP Network-wide Virtual Office login for one hour. This is known as Password guessing protection. When locked, an information message is written to the Call Server history file and printed to the TTY. The system administrator can unlock the TN by disabling and then enabling it in LD 32. This lockout does not survive re-registration of the IP Phone.

Virtual Office preemption

A user who is already logged into IP Network-wide Virtual Office can log in again on another IP Phone. This is useful if the user forgets to log out of an IP Phone and returns to their desk. The home IP Phone or the logged in IP Network-wide Virtual Office IP Phone is preempted upon a successful login on another IP Phone. The preempted IP Phone remains registered to the TPS but not to the Call Server, and is forced to register to the TN configured on the IP Phone.

Because only one registration can exist for a TN within a Call Server, the IP Phone is no longer operational, as indicated by the Logout Screen on the display.

The preempted IP Phone can be used by another user for a IP Network-wide Virtual Office login or it can be re-registered as the home IP Phone. The preempted Call Server registration may be a regular registration (one in which the IP Phone EEPROM S1 and TN settings match the login TN within the associated Call Server), an IP Network-wide Virtual Office login, or a Branch User login. If it is a regular registration, then the subsequent login is resolved within the Call Server to be a regular registration, not an IP Network-wide Virtual Office login or a Branch User login.

When a virtually logged in IP Phone resets, it attempts to register with the Call Server associated with its S1 and TN. If the TN has an IP Network-wide Virtual Office login or Branch User login, then it does not register to its 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 IP Phone 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.

Zone Based Dialing

If the Zone Based Dialing (ZBD) feature is enabled, then the following user ID formats for the Virtual Office (VO) login feature are valid or expected:

1. short DN (without numzone prefix).
2. zone specific ACx + 7_digit_DN.
3. system ACx + 7_digit_DN.

The system checks whether the dialed user ID matches with the ZFDP table for this numzone. If it is true, matched digits are deleted and the system validates the type of ZFDP rule.

Num.zone	Matching Digits	Type	Replacement digits	max length	Description
XX	<code digits>	LOC		LEN 16	Allow_VO_login_for_zone_XX

If this is LOC, system AC code is inserted. Otherwise (if no match is found, or it is not LOC rule), by default the user ID is appended with numzone prefix.

Possible valid VO user IDs are:

- XXXX as user ID and <numzone prefix> + XXXX DN exists, and have a valid configuration to VO login.
- <zone specific ACX> + <seven digits DN> (this DN can be from another numzone), ZFDP table have LOC rule with <zone specific ACX>.
- <system ACX> + <seven digits DN> (this DN can be from another numzone), ZFDP table have LOC rule with <system ACX>.

For more information on configuring Zone Based Dialing, see “Zone Based Dialing plan configuration” in ***Dialing Plans Reference, NN43001-283***.

Feature packaging

IP Network-wide Virtual Office requires the following packages:

- Virtual Office (VO) package 382
- Virtual Office Enhancement (VOE) package 387

Feature implementation

Task summary list

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

1. Table 75: LD 15 – Configuring Home Location Code (for UDP only) on page 248
2. Table 76: LD 11 – Configure IP Network-wide Virtual Office on page 249
3. Table 77: LD 86 – Configure Electronic Switched Network on page 249
4. Table 78: LD 87 – Configure the Local and Distant Steering Codes (for CDP only) on page 249
5. Table 79: LD 90 – Configuring the Access Code (for UDP only) on page 250

Table 75: LD 15 – Configuring Home Location Code (for UDP only)

Prompt	Response	Description
REQ	CHG	
TYPE	NET_DATA	Networking (NET).
CUST	xx	Customer number, as defined in LD 15.

Prompt	Response	Description
…		
- CLID	YES	Allow Calling Line Identification option.
- - SIZE	0-(256)-4000	CLID entry size
- - INTL	0-9999	Country code (1 to 4 digits)
- - ENTRY	xx	CLID entry to be configured
…		
HLOC	xx	Home Location Code.

The IP Network-wide Virtual Office feature does not support multiple HLOCs, LOCs, and Digit Manipulation blocks defined in LD 90.

Table 76: LD 11 – Configure IP Network-wide Virtual Office

Prompt	Response	Description
REQ:	NEW, CHG	
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
CUST	xx	Customer number, as defined in LD 15.
SCPW	xxxx	Station Control Password.
CLS	(VOLA) VOLD	Allow or deny IP Network-wide Virtual Office login operation on this TN.
CLS	(VOUD) VOUA	Allow or deny IP Network-wide Virtual Office user onto this TN using another phone.

Table 77: LD 86 – Configure Electronic Switched Network

Prompt	Response	Description
REQ	NEW, CHG	
CUST	xx	Customer number, as defined in LD 15.
FEAT	ESN	Electronic Switched Network (ESN).
…		
CDP	YES	Coordinated Dialing Plan feature.

Table 78: LD 87 – Configure the Local and Distant Steering Codes (for CDP only)

Prompt	Response	Description
REQ	NEW, CHG	
CUST	xx	Customer number, as defined in LD 15.

Prompt	Response	Description
FEAT	CDP	Coordinated Dialing Plan (CDP).
TYPE	LSC	Local Steering Code (LSC).
LSC	x…x	LSC.
…		
DSC	x…x	Distant Steering Code (DSC)

Table 79: LD 90 – Configuring the Access Code (for UDP only)

Prompt	Response	Description
REQ	CHG	
CUST	xx	Customer number, as defined in LD 15.
FEAT	NET	
TRAN	AC1	Translator.

You must also add the Gateway Endpoint and enable the Network Connection Server (NCS) flag on the Network Routing Service (NRS) . The Gateway Endpoint name must equal the H.323 ID for the Signaling Server as defined in the Element Manager. For more information about adding the Gateway Endpoint and enabling the NCS flag, see *Network Routing Service Installation and Commissioning, NN43001-564*. For more information about the H.323 ID for the Signaling Server in Element Manager, see *Element Manager System Reference—Administration, NN43001-632*.

You can also configure multiple routing entries for CDP and UDP. Depending on CDP configuration, configure the DN type to Private level 0 regional (CDP steering code). The DN prefix usually equals the leading digits of local DNs or LSC.

For the UDP dialing plan, configure the DN type to Private level 1 regional (UDP location code). The DN prefix equals or starts with HLOC, as defined in LD 15.

For more information about CDP, see *Dialing Plans Reference, NN43001-283*.

Feature operation

The IP Network-wide Virtual Office feature supports the following operations:

- IP Network-wide Virtual Office login on page 251
- IP Network-wide Virtual Office connection on page 255
- IP Network-wide Virtual Office logout on page 256

IP Network-wide Virtual Office login

IP Network-wide Virtual Office Login does not support Digit Manipulation or multiple Home Location Codes (HLOCs) as defined in BARS/NARS. If the HLOCs have common leading digits Virtual Office can support multiple HLOCs on the same Call Server. The IP Network-wide Virtual Office feature supports only one customer. If two or more customers are configured, the one with the lowest customer number is supported.

Because the IP Phone 2004 and IP Softphone 2050 have more key functions than the IP Phone 2002, a IP Network-wide Virtual Office login from an IP Phone 2002 to an IP Phone 2004 or IP Softphone 2050 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.

Because the IP Softphone 2050 does not support Digital Telephone Handsfree Denied (HFD) Class of Service:

- An IP Network-wide Virtual Office Login from an IP Softphone 2050 to an IP Phone 2001 is blocked.
- An IP Network-wide Virtual Office Login from an IP Softphone 2050 to an IP Phone 2002 or IP Phone 2004 with HFD Class of Service is blocked.

If a login is attempted in the above situation, an error message is displayed.

Features configured on DN/feature keys higher than key 3 on the IP Phone 2004, the IP Softphone 2050, or the MVC 2050are not accessible when logged in from an IP Phone 2002.

Use Performing IP Network-wide Virtual Office login on page 251 to log into IP Network-wide Virtual Office. The IP Phone is operating in Normal Mode.

Performing IP Network-wide Virtual Office login

1. Press the Services key to display the **Options** menu.

   The IP Phone 2001, IP Audio Conference Phone 2033, and IP Phone 2002 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 soft key.

   The screen prompts for the User ID.
4. Enter the User ID.

   The User ID is the user internal telephone number. Depending on the network configuration of the system and where the IP Network-wide Virtual Office login takes place, the User ID can be:

User ID	Explanation
UDP number	A user with a Home Location Code (HLOC) of 343 visits a site where the HLOC is 393. The User ID is 6 343 5555, where: 6 = network access code (access to ESN) 343 = Home Location Code (HLOC) 5555 = Directory Number (DN). HLOCs are defined in LD 15 Customer Data Block.
CDP number	A user with a HLOC of 343 wants to do an IP Network-wide Virtual Office login from the same site (HLOC = 343), on a different IP Phone. The User ID is 5555. The User ID of 6 343 5555 is also acceptable. HLOCs are defined in LD 15 Customer Data Block.
Transferable DN	A user with HLOC of 343, using a transferable DN, is permanently transferred to a site where HLOC is 393. The previous location DN (5555) is retained as the User ID. The User ID of 6 393 5555 is also acceptable.

5. Press the Select soft key.

   The screen prompts for the Station Control Password (SCPW).
6. Enter the SCPW for the Main Office IP Phone.

   When the User ID entered is not found locally, the message Locating Remote Server displays, indicating that connection setup is ongoing. A 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 IP Phone is redirected to the remote Home TPS.

IP Phone registered to the same Call Server

To log in to IP Network-wide Virtual Office from an IP Phone registered to the same Call Server, the User ID is the end user DN.

IP Phone registered to a different Call Server on the same Network

To log in to IP Network-wide Virtual Office from an IP Phone registered to a different Call Server on the same network, the User ID is comprised of the following:

- Access Code (AC1 or AC2 depending on which Access Code is used for the Uniform Dialing Plan [UDP])
- HLOC (defined in the Customer Data Block for the lowest numbered customer)
- DN of the end user

The following is an example of an IP Network-wide Virtual Office Login User ID using a 4-digit DN:

- AC1 is defined as 6
- HLOC is defined as 123
- DN is defined as 4567

Therefore, the IP Network-wide Virtual Office Login User ID is defined as 61234567.

The following is an example of an IP Network-wide Virtual Office Login User ID using a 5-digit DN:

- AC1 is defined as 6
- HLOC is defined as 12
- DN is defined as 34567

Therefore, the IP Network-wide Virtual Office Login User ID is defined as 61234567.

Login failures and errors

A IP Network-wide Virtual Office login fails if any of the following occurs:

- The login User ID is not local and there is a failure to get a response from a Gatekeeper.
- The User ID is not local and the Gatekeeper does not know the endpoint of the User ID.
- The User ID is not local and the Home TPS is unreachable.
- The User ID is not local and the IP Phone firmware required by the remote TPS node software is incompatible with that provided by the local TPS node.
- The User ID is not known or a MADN cannot be resolved at the Home Call Server.
- An SCPW is not configured for the User ID.
- The destination TN has Virtual Office User Denied (VOUD).
- There already exists a non-idle registered instance for the User ID.
- The user selects Cancel after a password failure.
- The password fails to authenticate after three attempts. The User ID entry in the Gatekeeper database points back to the originating Call Server.

See Table 80: System messages on page 254 for system messages, potential causes, and available actions.

Table 80: System messages

Message	Cause	Action
Busy, try again	Remote IP Phone is active (not idle).	Wait for remote IP Phone to become idle and try again.
	ACD is logged in.	Logout ACD before IP Network-wide Virtual Office logon from another IP Phone.
	Make-Set-Busy is inactive on ACD IP Phone.	Set Make-Set-Busy active on ACD IP Phone.
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 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 IP Phone in LD 32 at the Call Server to clear the lockout.
Logged Out	Home TN in use by IP Network-wide Virtual Office.	Logon to another TN using IP Network-wide Virtual Office. Re-register as the Home (or Branch) IP Phone.
Permission Denied (1)	SCPW is not configured or enabled.	Configure a SCPW for the remote IP Phone.
Permission Denied (3)	Incorrect User ID entered.	Enter correct User ID.
	SCPW is not configured.	Configure a SCPW for the remote IP Phone.
Permission Denied (4)	Incorrect User ID entered.	Enter correct User ID.
	Attempt to logon to an IP Phone 2004/IP Softphone 2050 TN from an IP Phone 2002.	Go to an IP Phone 2004/IP Softphone 2050 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 IP Phone with Virtual Office User Allowed (VOUA).

Message	Cause	Action
Permission Denied (6)	Incorrect User ID entered.	Enter correct User ID.
	Incorrect SCPW is entered.	Enter correct SCPW.
Permission Denied (7)	Attempted Login to an IP Phone 2001, or IP Audio Conference Phone 2033 from an IP Softphone 2050. Attempted Login to an IP Phone 2002 or IP Phone 2004 with CLS HFD from an IP Softphone 2050.	Go to an IP Phone 2001, IP Audio Conference Phone 2033, IP Phone 2002, or IP Phone 2004 and try again. Go to an IP Phone 2002 or IP Phone 2004 and try again.
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.	Restore remote TPS.
	Link to remote TPS is down.	Restore link to remote TPS.
	Remote Call Server is down.	Bring remote Call Server up.

IP Network-wide Virtual Office connection

An IP Phone is registered with the TN in its EEPROM and then a User ID and password are used to determine the home TPS for the IP Phone during the IP Network-wide Virtual Office connection. A Nortel Succession Gatekeeper is required, if the home TPS is not the TPS where the IP Phone is registered when the user initiates an IP Network-wide Virtual Office login.

Upon logon, the IP Phone receives the features, time, date, and tones of the home Call Server. The IP Phone becomes part of the home zone and receives call service from this Call Server. When this zone is remote the voice quality may be less than ideal because codec selection is based on the IP Phone being in its expected location (subnet). Firmware upgrade on a TPS node does not upgrade the telephones logged into IP Network-wide Virtual Office.

The IP Phone options stored locally in the telephone non-volatile memory retain the characteristics of the IP Phone. That is, items from the Option menu such as language, display contrast, date and time format, and ring type retain the IP Phone properties and do not change to reflect the preferences on the IP Network-wide Virtual Office user home IP Phone.

If the IP Network-wide Virtual Office user changes these settings, the changes persist in the IP Phone even after logout, power-up, or reboot. In other words, changes to Telephone Options menu items are retained (in the IP Phone 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 Call Server through IP Network-wide 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.

IP Network-wide Virtual Office logout

The IP Network-wide Virtual Office user initiates a logout by:

- Performing a direct IP Network-wide Virtual Office logout on the active IP Phone (see Performing IP Network-wide Virtual Office logout on page 256).
- Returning to the home location and re-registering the home IP Phone (using the Home softkey). This forces a logout on the idle IP Network-wide Virtual Office IP Phone registered to the same User ID.
- Performing an IP Network-wide Virtual Office login on another IP Phone. This forces a logout on the idle IP Network-wide Virtual Office IP Phone registered to the same User ID.

The IP Network-wide Virtual Office user is forced to logout when:

- the IP Phone power cycles
- the IP Phone loses connectivity to the Home TPS
- the automatic logout activates (see IP Network-wide Virtual Office auto-logout on page 257)

Upon logout, the IP Phone returns to its regular keymap unless the TN owner has an active IP Network-wide Virtual Office login or Branch User login. In these cases, the IP Phone displays the Logout screen. If connectivity to the Home Office is lost during an IP Network-wide Virtual Office login, calls are not maintained and the IP Phone re-registers to its configured TN and Call Server.

Use Performing IP Network-wide Virtual Office logout on page 256 to logout of IP Network-wide Virtual Office. The IP Phone is operating in Normal Mode.

Performing IP Network-wide Virtual Office logout

1. Press the Services key to display the **Options**menu.

The IP Phone 2002, IP Phone 2001, IP Audio Conference Phone 2033 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 soft key.

IP Network-wide Virtual Office auto-logout

The IP Network-wide Virtual Office user can be configured to automatically logout all IP Network-wide 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.

For more information on prompts and responses in LD 15, see *Software Input Output - Administration, NN43001-611*.

Chapter 28: IP Phone 1210 Last Number Redial soft key

Contents

This section contains information about the following topics:

Feature description

Most IP Phone users use the Last Number Redial (LNR) feature by pressing the LNR feature key or by pressing the DN key twice. Since the IP Phone 1210 does not have programmable feature keys or a DN key, this feature provides the IP Phone 1210 with a LNR soft key. Add or remove this soft key by configuring the Last Number Redial Allowed (LNA) or Last Number Redial Denied (LND) Class of Service (CLS).

The last number redial soft key appears after the user takes the IP Phone 1210 off-hook or presses the Handsfree key. The Call Server sends the LNA or LND CLS during registration or after you make configuration changes.

Operating parameters

You do not need to restart the IP Phone 1210 after you make configuration changes to the LNA or LND CLS.

Only the IP Phone 1210 uses the LNR soft key.

The IP Phone 1210 translates the LNR soft key to all available languages.

After an IP Phone 1210 user presses the Handsfree key, the Last# soft key also appears. Configure Handsfree Allowed (HFA) and LNA CLS to add the Last# key. If you add or remove other soft keys, the location of the Last# key automatically updates.

Feature interactions

The IP Phone 1210 LNR soft key provides all functionality of the basic LNR feature.

Feature packaging

The IP Phone 1210 Last Number Redial soft key requires Last Number Redial (LNR) package 90.

Feature implementation

Use Element Manager to configure LNR on the IP Phone 1210.

Configuring LNR CLS

1. Log on to Element Manager with a valid user account.
2. In the Navigator pane, select **Phones**.

   The Search for Phones Web page appears.
3. Select a search criteria from the Criteria list.
4. Sort the telephone list by telephone type, and then click the box beside the telephones to update.

   The **Phone Details** Web page appears.
5. Scroll to the Features section.
6. For LNR, select **LNA** from the list.
7. Click Save.

Feature operation

If you configure the LNA CLS and the IP Phone user takes the IP Phone 1210 off-hook, the Last# soft key appears in the list of context-sensitive soft keys. The IP Phone user presses the Last# soft key to automatically redial the last number dialed.

If Handsfree (HFA) CLS is configured for the IP Phone 1210, the Last# soft key appears after the IP Phone user presses the Handsfree key. The IP Phone user presses the Last# key to automatically redial the last number dialed.

If LND CLS is configured for the IP Phone 1210 and the IP Phone user takes the IP Phone 1210 off-hook or presses the Handsfree key, the Last# key does not appear in the list of context -sensitive soft keys.

Chapter 29: IP Phone Audio Message Waiting Indication

Contents

This section contains information about the following topics:

Feature description

Use this feature to support audio-based Message Waiting Indication (MWI) on an IP Phone. Prior to this feature, only an LED on the IP Phone notified the IP Phone user of a message waiting. Use this feature to provide both audio-based and LED-based notification.

Configure audio-based MWI by using the Message Waiting Tone Allowed (MWTA) Class of Service (CLS).

Operating parameters

Configure the Message Waiting Tone Allowed (MWTA) CLS. After the IP Phone user receives a message, the IP Phone user hears a message-waiting interrupted dial tone on the hands-free or handset speaker.

Feature interactions

The Message Waiting Tone feature does not work if Message Intercept Allowed (MINA) is configured.

Feature packaging

This feature is included in base system software.

Feature implementation

Use Element Manager to configure audio-based MWI on IP Phones.

Configuring MWTA CLS

1. Log on to Element Manager with a valid user account.
2. In the Navigator pane, select **Phones**.

   The Search for Phones Web page appears.
3. Select search criteria from the Criteria list.
4. Sort the telephone list by telephone type, and then click the check box beside the telephone to update the list.

   The **Phone Details** Web page appears.
5. Use the scroll bar to navigate to the Features section.
6. For MWTA, select **Allowed** from the list.

   See Figure 12: MWTA on Phone Details Web page on page 265.

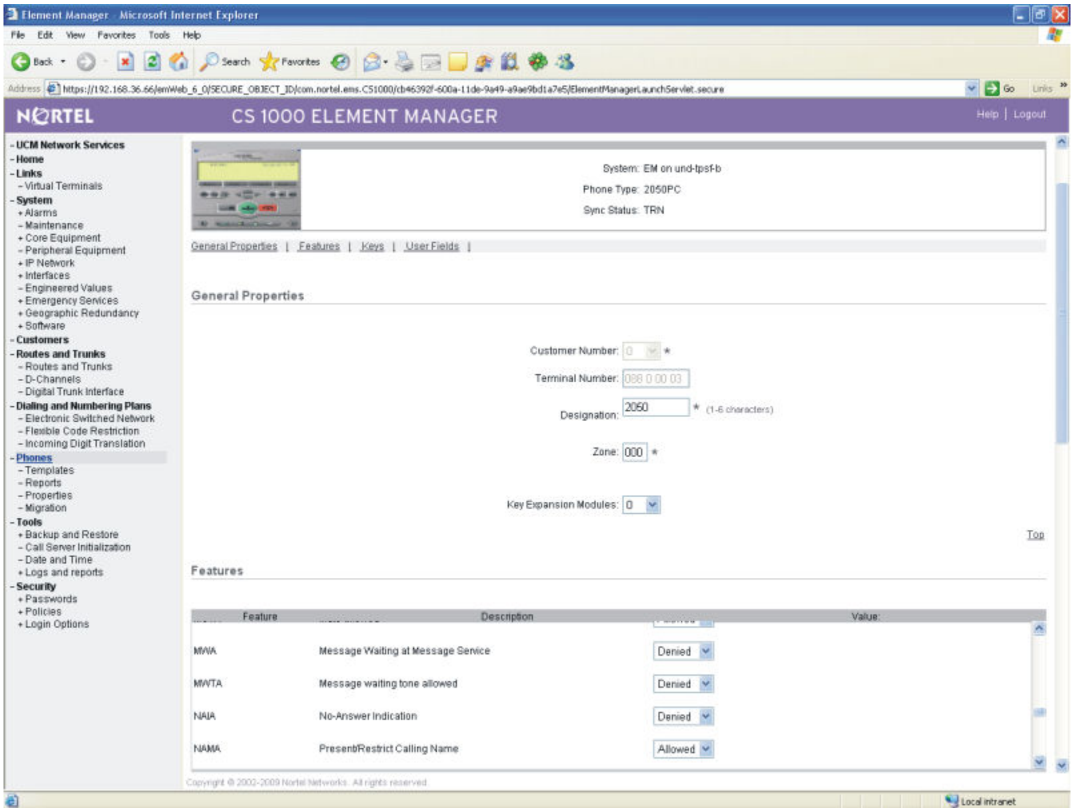


Figure 12: MWTA on Phone Details Web page

Feature operation

No specific operating procedures are required to use this feature.

Chapter 30: IP Phone Disable Mute function

Contents

This section contains information about the following topics:

Feature description

This feature allows administrators to disable the mute function of IP Phones that have a Mute key. If the mute function is disabled, then pressing the Mute key places the active call on hold; pressing the Mute key again takes the held call off hold.

Two classes of services are available.

- MUTA (Mute Key Allowed) class of service for IP Phone. If the IP Phone user presses the Mute key on the IP Phone during an established call, then a one-way speech path is established and the call on the second telephone cannot hear the speech path. If the user presses the Mute key on the IP Phone again, then the two-way speech path is re-established.
- MUTD ((Mute Key Denied) class of service for IP Phone. If the user presses the Mute key on the IP Phone during an established call, then the call is placed on hold; the other party may hear music on hold if music is configured. To take the call off hold, the IP Phone user can press the Mute key again, or press the DN key.

Note:

If the IP Phone user places the call on hold using the Hold key, the user must retrieve the held call by pressing the Hold key again; pressing the Mute key does not take the call off hold.

Operating parameters

This feature is available only for IP Phones with a Mute key:

- IP Phone 2002
- IP Phone 2004
- IP Phone 2007
- IP Audio Conference Phone 2033
- IP Softphone 2050
- Mobile Voice Client 2050
- IP Phone 1120E
- IP Phone 1140E
- IP Phone 1150E
- IP Phone 1165E
- IP Phone 1210
- IP Phone 1220
- IP Phone 1230

MUTD class of service is applicable for each Directory Number (DN) key configured for the phone.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Use Element Manager to configure audio-based MWI on IP Phones.

Configuring MUTA CLS

1. Log on to Element Manager with a valid user account.
2. In the Navigator pane, select **Phones**.

   The Search for Phones Web page appears.
3. Select search criteria from the Criteria list.
4. Sort the telephone list by telephone type, and then click the check box beside the telephone to update the list.

   The **Phone Details** Web page appears.
5. Use the scroll bar to navigate to the **Features** section.
6. For MUTA, select **Allowed** from the drop-down list.

   See Figure 13: Configure MUTA/MUTD class of service in Element Manager on page 269.

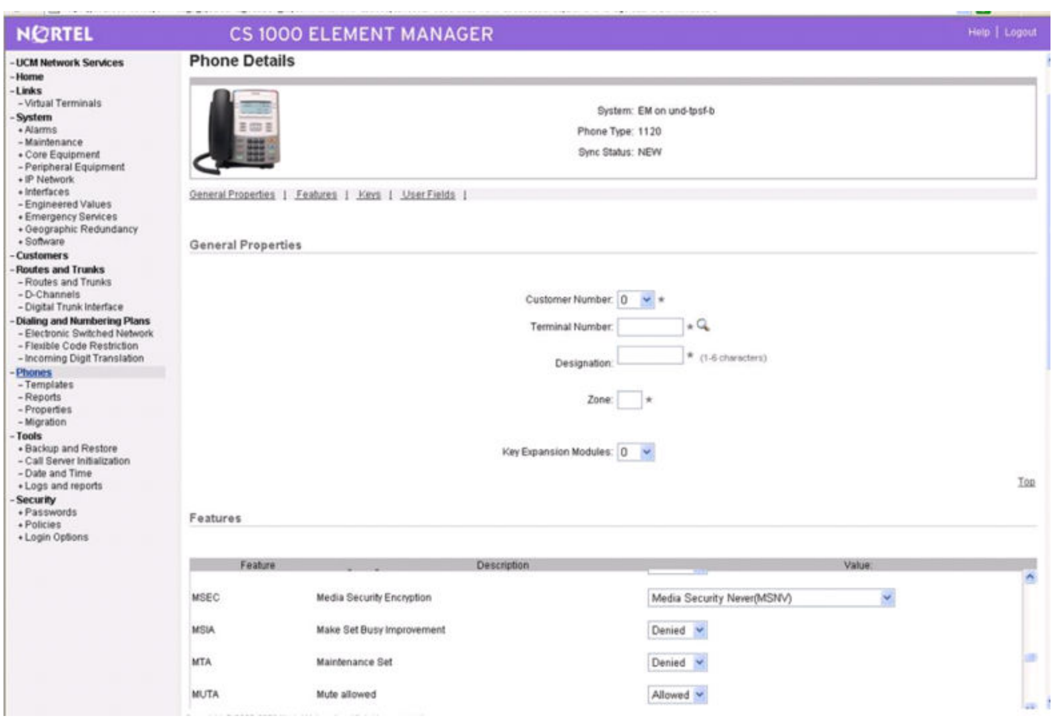


Figure 13: Configure MUTA/MUTD class of service in Element Manager

Feature operation

If MUTA is configured, press the Mute key on the IP Phone during an established call to establish a one-way speech path; the call on the second telephone cannot hear the speech path. Press the Mute key on the IP Phone again to re-establish the two-way speech path.

If MUTD is configured, press the Mute key on the IP Phone during an established call to place the call on hold. To take the call off hold, press the Mute key again, or press the DN key.

Chapter 31: IP Phone Password Protection for Language and Feature Key Labels

Contents

This section contains information about the following topics:

Feature description

Use this feature to configure password protection for language and feature key labels. You can configure this feature for new or existing IP Phones. If you configure Controlled Class of Service Allowed (CCSA) and Station Control Password (SCPW), the IP Phone prompts for the SCPW before the IP Phone user gains entry to the Language menu or the Feature Key Label menu.

Operating parameters

Use this feature only for IP Phones.

You must configure CCSA Class of Service (CLS) and define the SCPW .

Feature interactions

IP Phone Password Protection for Language and Feature Key Labels menu access depends on the existing Controlled Class Of Service (CCOS) functionality.

Feature packaging

IP Phone Password Protection for Language and Feature Key Labels requires Controlled Class of Service (CCOS) package 81.

Feature implementation

Task summary list

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

1. Configure CCSA in Element Manager on page 272
2. Define SCPW in Element Manager on page 273

Configure CCSA in Element Manager

1. Log on to Element Manager with a valid user account.
2. In the Navigator pane, select **Customers**.

   The Customers Web page appears.
3. Click the **Customer Number**.
4. Click **Edit**.

   The Edit Web page appears.
5. Click **Controlled Class of Service**.

   The Click Controlled Class of Service Web page appears.
6. Enter the appropriate information, and then click **Save**.

   See Figure 14: Controlled Class of Service Web page on page 273.

Controlled Class of Service Web page

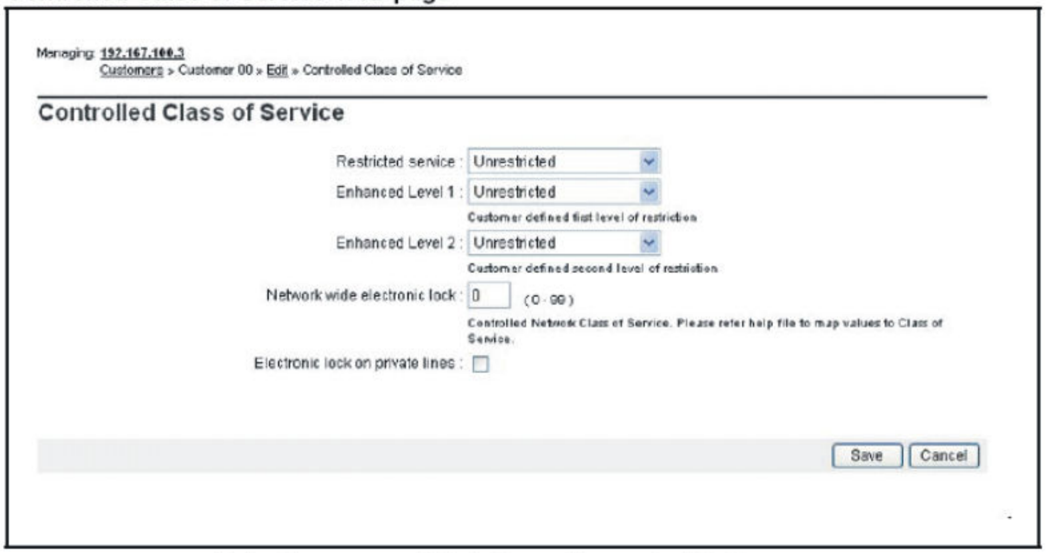


Figure 14: Controlled Class of Service Web page

Define SCPW in Element Manager

1. Log on to Element Manager with a valid user account.
2. In the Navigator pane, select **Phones**.

   The Search for Phones Web page appears.
3. Select a search criteria from the Criteria list.
4. Sort the telephone list by telephone type, and then click the box beside the telephones to update.

   The **Phone Details** Web page appears.
5. Scroll to the Features section.
6. For SCPW, enter appropriate password and then click Save.

Feature operation

If you enable this feature, the IP Phone user must enter a valid password to change language or feature key labels. The password prompt appears when the IP Phone user tries to enter the Language or Feature Key Label menus.

If the IP Phone user enters an invalid password, an invalid password message appears and prompts the IP Phone user to enter the password again. If the IP Phone user enters an invalid password more than three times, the IP Phone locks the password prompt.

If you do not enable this feature, the IP Phone user has unrestricted access to Language and Feature Key Label menus.

If you configure CCSA CLS but do not define SCPW, the message `Password undefined` appears after the IP Phone user enters the Language or Feature Key Label menus. If you configure CCSA or CCSD CLS while an IP Phone user is in the Language or Feature Key Label menus, the configuration change takes effect after the IP Phone user exits the menu.

Chapter 32: IP Phone single-line-display of PD, CL, RL, and Corporate Directory additional information

Contents

This section contains information about the following topics:

Feature description

The single-line-display IP Phones include the IP Phone 2002, IP Phone 1120E, IP Phone 1220, and IP Phone 1230.

This feature enables the IP Phone user to scroll Personal Directory/Redial List/Callers List (PD/RL/CL) records by DN and switch between CARD and LIST views in the Corporate Directory.

On these IP Phones, the summary screen shows information about PD/RL/CL in general. The List Screen shows the list of names, and the view screen displays a list of DNs. An additional state, the browse screen, can display user/call information for single-line-display IP Phones. The navigation buttons (Up/Down/Right/Left) are used to switch between these states. See Figure 15: Single-line-display IP Phones on page 276.

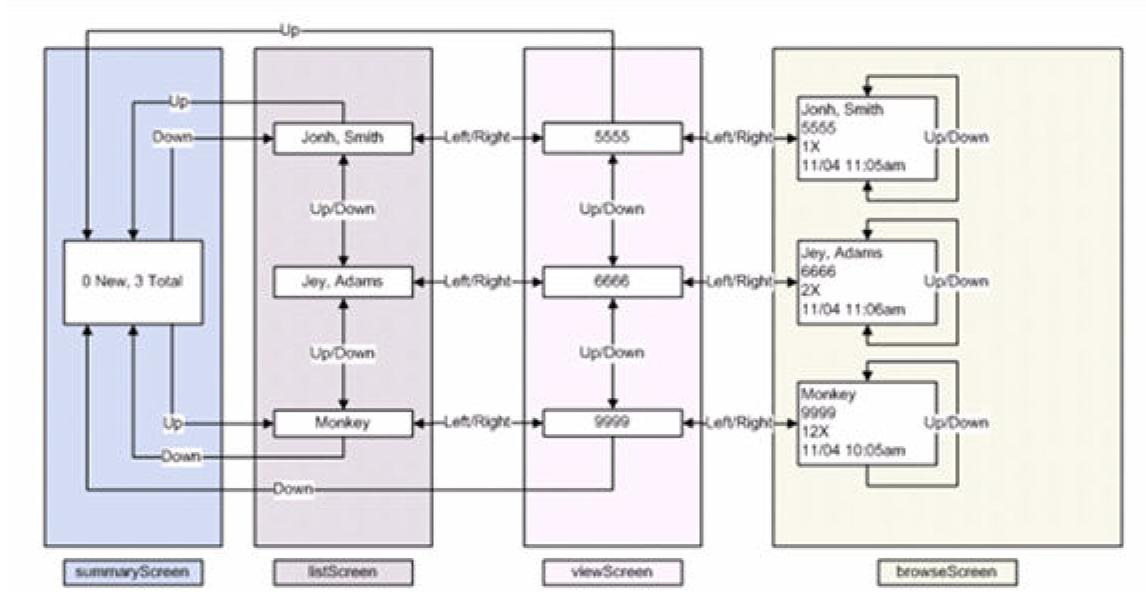


Figure 15: Single-line-display IP Phones

Corporate directory switches between two states: CDIR_LIST_MODE and CDIR_CARD_MODE. In CDIR_LIST_MODE, Corporate Directory shows a list of records arranged by name; only names are shown. In CDIR_CARD_MODE name, the DN and organization are shown for each record. On single-line-display IP Phones, the Up/Down keys are used to scroll through one record. See Figure 16: Corporate Directory on page 276.

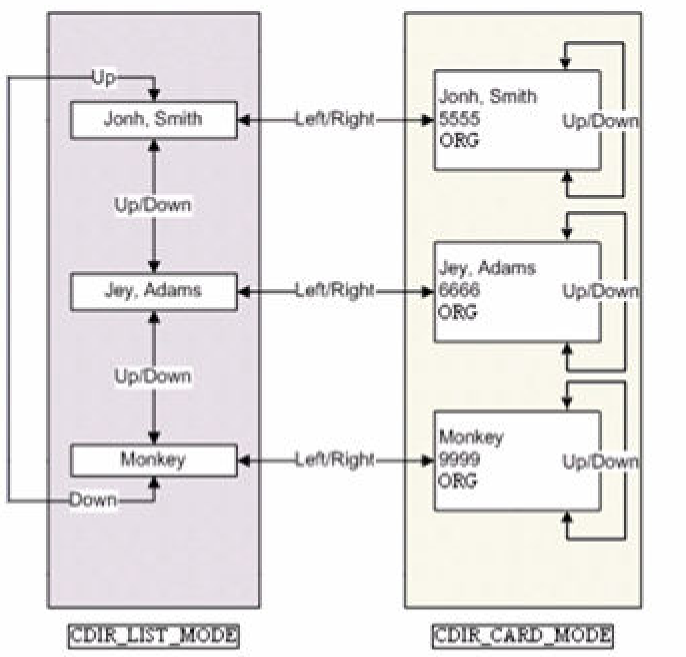


Figure 16: Corporate Directory

Operating parameters

This feature is applicable to the IP Phone 2002, IP phone 1120E, IP Phone 1220, and IP Phone 1230.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Use LD 11 to configure this feature.

Table 81: LD 11 Additional information display for single-line-display IP Phones

Prompt	Response	Description
REQ	CHG	Change
TYPE	aaaa	Telephone type
TN	l s c u	Terminal number
ECHG	YES	Easy Change
CLS	CNDA DNDA CRPA	Call Number Display Allowed Dialed Name Display Allowed Corporate Directory Allowed

Feature operation

Use the navigation buttons (Up/Down/Right/Left) to switch between the use screen and browse screen states in PD/CL/RL.

For Corporate Directory, in CDIR_CARD_MODE, the DN and organization are shown for each record. Use the Up/Down keys to scroll through one record.

Chapter 33: IP Recorded Announcements

Contents

This section contains information on the following topics:

Feature description

IP Recorded Announcements provides the same functionality and features as TDM Recorded Announcements. The Recorded Announcement (RAN) feature allows the system to connect calls automatically to a customer-provided Recorded Announcement machine.

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.

Trunk units for IP RAN route data blocks are configured on virtual superloops using the same overlays used to configure traditional RAN trunks. The density of the IP RAN card is extended to 32 units, so you can configure units 0 to 31.

The IP RAN broadcast feature allows multiple calling parties to receive RAN service from one RAN trunk. You can configure the maximum number of connections that a RAN trunk can accept. The system selects a registered IP RAN trunk for service delivery until the maximum number of connections is reached. Then it selects a new IP RAN trunk.

IP RAN is supported for all IP Phones and firmware versions.

For more information about Recorded Announcements, see the entry for Recorded Announcements in *Features and Services Fundamentals, Book 5, NN43001-106*.

Operating parameters

Before you can use this feature, you must configure the Media Application Server (MAS) and Network Routing Service (NRS) for IP Media Services. For information about configuring MAS and NRS for IP Media Services, see *Signaling Server and IP Line Applications Fundamentals (NN43001–125)*.

The IP Media Sessions license reflects the overall number of possible IP sessions with MAS. It is the sum of IP Music + IP Announcement + IP Ad Hoc Conference + IP Attendant Console + IP Tone. The IP RAN license works like the traditional digital RAN license. It is based on the number of broadcast IP RAN connections available on the system. As each new broadcasting RAN trunk is configured, the number of available broadcast connections subtracts from the maximum number of broadcast connections to the IP RAN trunk (4-48, depending on the CONN prompt value for the trunk). IP Media Services licenses are decremented simultaneously.

For more information about operating parameters, see the entry for Recorded Announcements in *Features and Services Fundamentals, Book 5, NN43001-106*.

Feature interactions

For more information about feature interactions, see the entry for Recorded Announcements in *Features and Services Fundamentals, Book 5, NN43001-106*.

Feature packaging

The IP RAN application is included with IP Media Services Package 422.

Feature implementation

Trunk units for IP RAN route data blocks are configured on virtual superloops using the same overlays used to configure traditional RAN trunks. The density of the IP RAN card is extended to 32 units, so you can configure units 0 to 31.

Note:

If you are configuring IP RAN and IP Music trunks on the same card number in the system, you must keep the two types separated from each other in blocks of 4 units. For example, if you configure IP RAN trunks in units 0-3, the system does not permit the configuration of IP Music trunks in the same group of 4 units.

Overlays

Task summary list

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

- Table 82: LD 16 - Create or modify IP RAN route data block on page 281
- Table 83: LD 14 - Configure a trunk for an IP RAN route data block on page 282

Table 82: LD 16 - Create or modify IP RAN route data block

Prompt	Response	Description
REQ	aaa	NEW or CHG
TYPE	RDB	Route Data Block
CUST	xx	Customer number, as defined in LD 15
ROUT	x...x	Route number
...		
TKTP	IRAN	IP RAN trunks
ZONE	0-255	Zone for codec selection and bandwidth management
NODE	xxxx	Node ID
...		
RTYP	MAS	MAS is automatically selected for IRAN route data block types
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).

Prompt	Response	Description
	DIS	IRAN 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
BDCT	YES (NO)	Enable broadcast capability for this route. **Note:** For CS 1000E systems, the default is YES. CS 1000E systems only support broadcast trunks.
ASUP	(NO) YES	Supervision is (or is not) required to inform Central Office (CO) when call is answered.
ACOD	xxx...x	Trunk route access code

Table 83: LD 14 - Configure a trunk for an IP RAN route data block

Prompt	Response	Description
REQ	NEW	New
TYPE	IRAN	IP Recorded Announcements trunk
TN	l s c u	Terminal number
DES	x...x	Designator field, where x...x can be a string of 0-16 alphanumeric characters
XTRK	IPMS	Extended trunk
CUST	xx	Customer number, as defined in LD 15
RTMB	xxx xxxx	Route number and Member Number
CONN	(4)-48	Maximum number of broadcast connections allowed for this trunk

Table 84: LD 117 - Check status of IP Announcement

Command	Description
STAT SERV APP IPANN	Prints the status of servers that have IP Announcement listed as an application.

Element Manager

IP RAN trunks and routes can be configured using the **Routes and Trunks** branch of the Element Manager navigator.

Configuring RAN route and trunk using Element Manager

1. In the Element Manager navigation tree, click the **Routes and Trunks** link.

   The New Route Configuration page appears, as shown in the following figure.

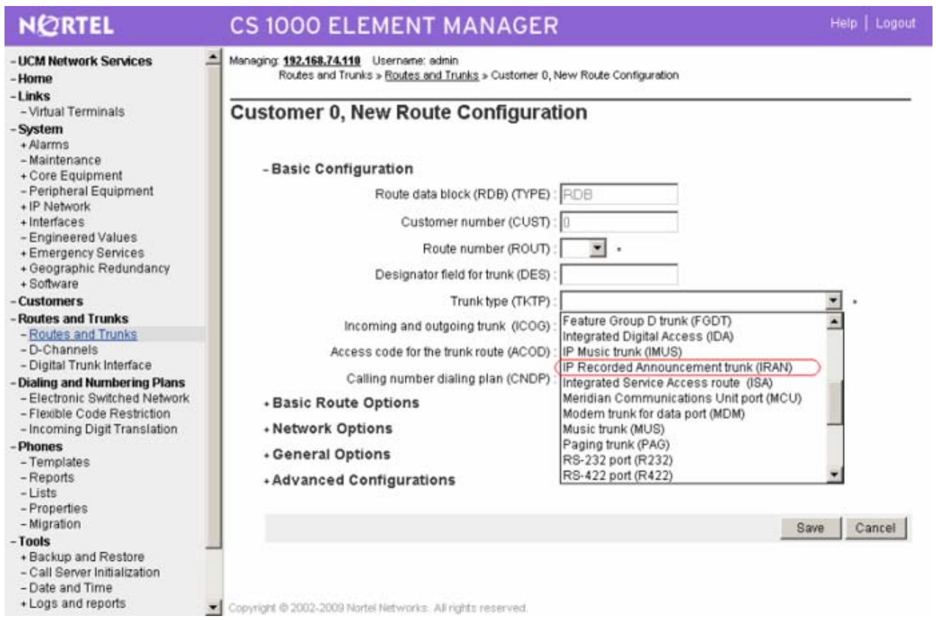


Figure 17: New Route Configuration page

2. In the Basic Configuration section, select **IP Recorded Announcement Trunk (IRAN)** for the Trunk type.

   The virtual trunk route (VTRK) option is automatically selected when creating an IP RAN route, as shown in the following figure.

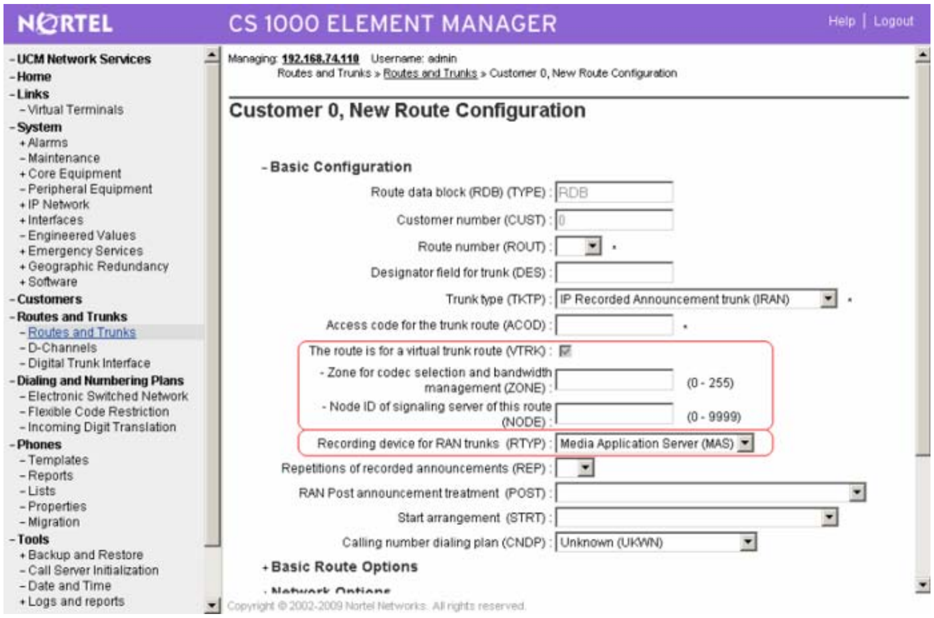


Figure 18: VTRK selected for IP RAN route trunk type

3. Configure the route properties and click **Save**.

   The Property Configuration page appears, as shown in the following figure.

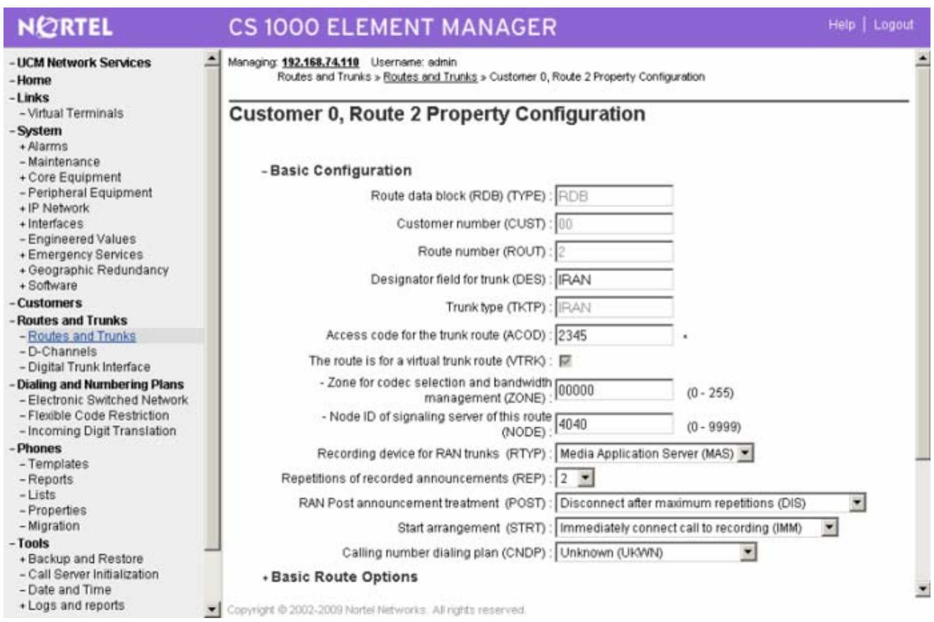


Figure 19: Property Configuration page

4. Configure the route properties and click **Save**.

   The New Trunk Configuration page appears, as shown in the following figure.

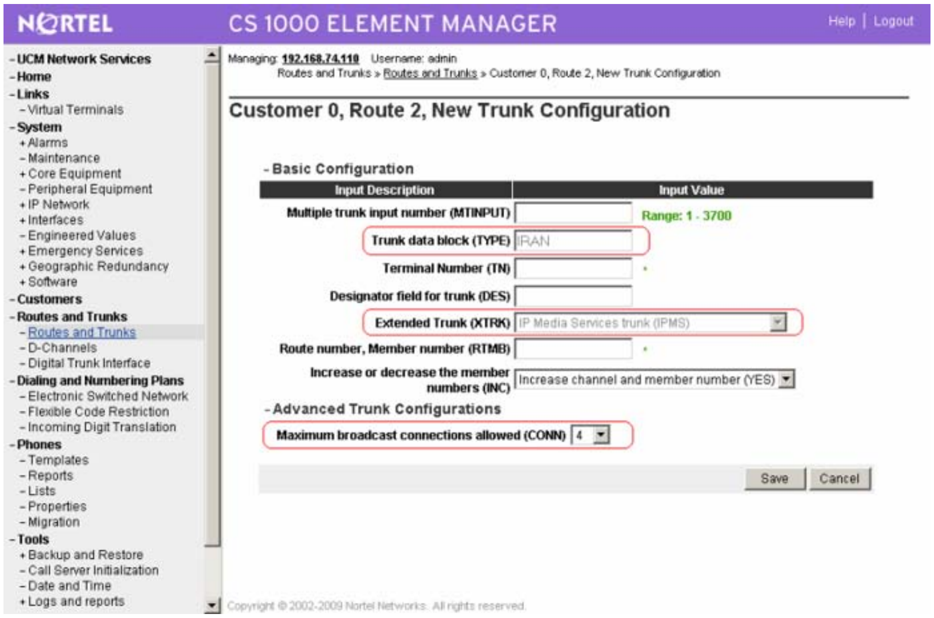


Figure 20: New Trunk Configuration page

The Trunk data block type is IRAN and the Extended Trunk (XTRK) is configured as IP Media Services trunk (IPMS). These values cannot be changed.

5. Select the maximum number of broadcast connections to be allowed.
6. Click **Save**.

For more information about Element Manager, see *Element Manager System Reference - Administration, NN43001-632*.

Chapter 34: IP Tone

Contents

This section contains information on the following topics:

Feature description

IP Tone provides the same functionality as traditional Tone. However, IP tone loops are selected for IP endpoints only. This includes IP Phones, IP Conference, IP Announcement, virtual trunks (SIP and H.323), and all devices connected using virtual trunks (such as SIP DECT and SIPL, and so on). If an IP Tone loop is not available, then a TDS loop is selected. However, if the preferred service is TDS and there are no TDS loops available, an IP Tone loop is not used.

IP Tone is supported for all IP Phones and firmware versions.

Survival Tones

In High Scalability systems, the Survivable SIP Media Gateway (Survivable SIP MG) consists of a Survivable Call Server (SCS) and SIP Media Gateway (SIP MG) that provide IP resources in the event of WAN outage. During a WAN outage, IP Phones register locally to the SCS. To provide call progress tone (ring back tone only), tone service is requested from the SIP MG using the same method that the IP Media Services applications use to request tone service from the MAS, provided that the SCS is running the IP Media Services application and the SIP MG IP address is configured as the local media server for IP Media Services. When the SIP MG receives the tone request, the SIP Gateway application translates the request to an ACD

DN call. The Call Server places the call into an ACD queue and provides in-band ring back tone to the call originator.

Operating parameters

The IP Media Sessions license reflects the overall number of possible IP sessions with MAS. It is the sum of IP Music + IP Announcement + IP Ad Hoc Conference + IP Attendant Console + IP Tone. The IP Tone licenses are controlled by the IP Media Services licenses. Available licenses do not decrement until an IP Tone loop is used. They are decremented by 1 for each loop channel used.

Before you can use this feature, you must configure the Media Application Server (MAS) and Network Routing Service (NRS) for IP Media Services. For information about configuring MAS and NRS for IP Media Services, see *Signaling Server and IP Line Applications Fundamentals (NN43001–125)*.

Feature interactions

For information about feature interactions, see the entry for Tones and Cadences in *Features and Services Fundamentals, Book 6, NN43001-106*.

Feature packaging

The IP Tones feature is included with IP Media Services Package 422.

Feature implementation

You can add, modify, remove, or check the status of IP conference loops using the overlay commands described in the following section.

Overlays

Task summary list

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

- Table 85: LD 17 - Configure a virtual IP Tone loop on page 289
- Table 86: LD 34 - Enable or disable IP Tone loop on page 289
- Table 87: LD 117 - Print status of IP Tone application on page 290

Table 85: LD 17 - Configure a virtual IP Tone loop

Prompt	Response	Description
REQ	xxx	NEW or CHG
TYPE	CEQU	Common Equipment parameters
...		
IPTONE	0-255	Virtual IP Tone loop You can add multiple tone loops at the same time. Precede a tone loop number with X to remove it. You can remove multiple tone loops at the same time. **Note:** You must first disable a tone loop before removing it.
NODE	1-9999	Node ID of the IP Tone loop.

Table 86: LD 34 - Enable or disable IP Tone loop

Command	Description
ENLL <loop>	Enable the specified IP tone loop
DISL <loop>	Disable the specified IP tone loop
STAT <loop>	Print the status of the specified IP tone loop

Command	Description
	Output format for IP tone loop is: • IPTONE n DSBL n BUSY n REG = number of IP tone groups disabled and busy and the registration status of the IP Media Services Conference Controller, where: - 00 = IP Media Services Controller is not registered with the Call Server - 30 = Maximum number of IP tones are registered and available for use

Table 87: LD 117 - Print status of IP Tone application

Command	Description
STAT SERV	Prints the status of all servers registered to the Call Server.
STAT SERV APP IPTONE	Prints only the status of servers that have IP Tone listed as an application.

Element Manager

Use the following procedure to configure an IP Tone loop using Element Manager.

Configuring an IP Tone loop using Element Manager

1. To configure or edit Loops information, click the **Core Equipment > Loops** link of the System branch of the Element Manager navigator.

   The Loops Web page appears.

2. From the Select a loop list menu, select IP TDS Loop.

   The IP TDS Loop Number Details page appears.

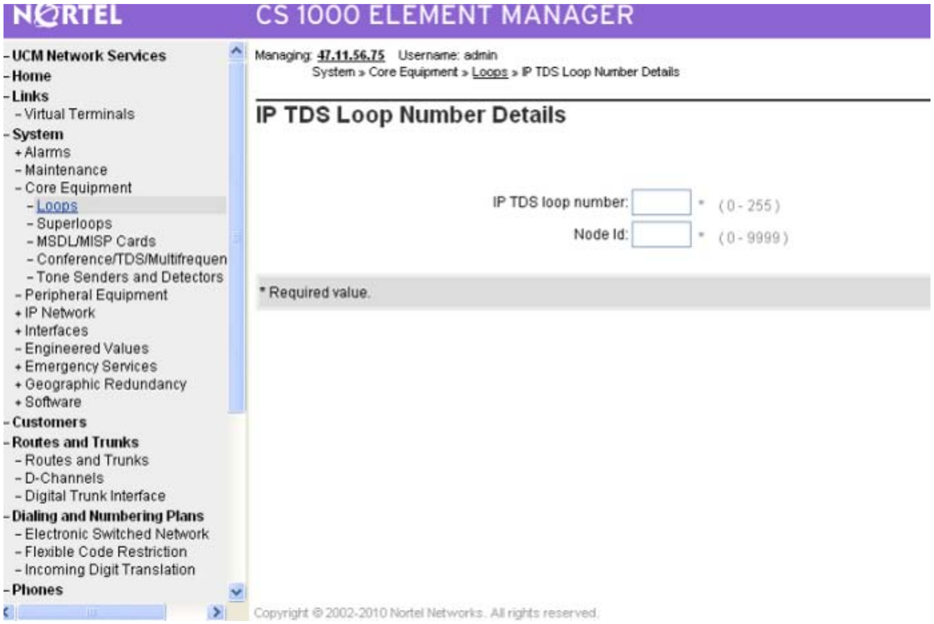


Figure 21: IP TDS Loop Number Details page

3. Enter the IP TDS loop number. The value can be 0-255.
4. Enter the Node ID. The value can be 0-9999.
5. Click **Save**.

Chapter 35: Italian Central Office Special Services

Contents

This section contains information on the following topics:

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 telephone accessing a special "1xx" service cannot establish a conference by pressing the Conference key or Loop key.

Analog trunks on Small 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 telephone 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

Table 88: 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.

Chapter 36: Italian Periodic Pulse Metering

Contents

This section contains information on the following topics:

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 is 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 because 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

Table 89: 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.

Chapter 37: KD3 Direct Inward Dialing and 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 is supported (no other networking protocols are 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, see the KD3 Signaling document contained in the IPE supplement for Spain.

Chapter 38: Last Number Redial

Contents

This section contains information on the following topics:

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 M2317 telephone has 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, 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.

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 are 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 is 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 is 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.

Speed Call, System

A number dialed using a System Speed Call key becomes the Last Number Redial number on all telephones, except the M2317. 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. Table 90: LD 15 on page 308

   Enable or disable LNR for a customer.
2. Table 91: LD 10 on page 308

   Add or change LNR for analog (500/2500-type) telephones.
3. Table 92: LD 11 on page 309

   Add or change LNR for Meridian 1 proprietary telephones.

Table 90: LD 15

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system
	0-31	Range for Small System and Media Gateway 1000B
- OPT	(LRD) LRA	LNR (denied) allowed

Table 91: LD 10

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

Prompt	Response	Description
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit
CLS	(LND) LNA	LNR (denied) allowed
LNRS	4-(16)-31	LNR size

Table 92: LD 11

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit
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.

Chapter 39: Limited Access to Overlays

Contents

This section contains information on the following topics:

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 limited to:
    - no access at all to LD 17
    - changing a user 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, see LD 17, prompts PWD2, NPW1, NPW2, and LD 15, prompts ATAC and SPWD in *Software Input Output - Administration., NN43001-611*.

Each password can access up to 32 customer-tenant combinations. Each combination is defined by a number designator that includes the customer number (0 to 99) and the tenant number (0 to 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 logon 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)

```
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
```

Figure 22: Example of Audit Trail printout (LD 22)

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 logon attempts for Administrator or Installer sets. Lockout procedures are not used.
- TTY users are not permitted to logon 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 telephone 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 telephone 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. Table 93: LD 17 on page 315

   Define LAPW options and passwords.
2. Table 94: LD 17 on page 316

   Change user LAPW password (user must log in using current LAPW).
3. Table 95: LD 22 on page 316

   Check options available for LAPW passwords (administrator).
4. Table 96: LD 22 on page 317

   Print options for LAPW password (user).
5. Table 97: LD 22 on page 317

   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.

Table 93: LD 17

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.

Prompt	Response	Description
		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.

Table 94: LD 17

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.

LAPW password options are output to the active TTY only. Options format is shown below:

Table 95: LD 22

Prompt	Response	Description
REQ	PWD	Lookup password options.
PWD2	xxxx	Level 2 master password.
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.

Prompt	Response	Description
HOST	No	Host mode.
OPT	aaaa . . .	Password options allowed.

Options available to the logged on password are printed. The format is shown below:

Table 96: LD 22

Prompt	Response	Description
REQ	PWD	Print passwords.
PWD2	<CR>	Administrator password.
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.

Table 97: LD 22

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.

Chapter 40: Limited Access to Overlays Password Enhancement

Contents

This section contains information on the following topics:

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

The OPT prompt associated with LAPW Password Option Access Rights accepts the access rights for Loss Planning Customization Allowed (LOSA) or Denied (LOSD).

Table 98: LD 17

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.

Prompt	Response	Description
- - 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.

Chapter 41: Line and Trunk Cards

Contents

This section contains information on the following topics:

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.

- Digital Line Cards
    - NT8D02 Digital Line Card (16 digital telephones plus 16 associated data options/attendant console)
    - NTDK16 Digital Line Card (Chassis systems only; 48 telephones plus 16 associated data options/attendant console)
- analog (500/2500-type) telephone Line Cards
    - NT1R20 Off-Premise Station Analog Line Card
    - NT8D03 Analog Line Card (16 analog in-line telephones)

- NT8D09 Analog Message Waiting Line Card (16 analog single-line telephones with Message Waiting lamps)

In addition, Data Line Cards are available to interface data communication products.

For information on ITG line cards, see *Signaling Server IP Line Applications Fundamentals, NN43001-125*.

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)

For information on ITG Trunk cards, see *IP Trunk Fundamentals, NN43001-563*.

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.

Chapter 42: Line Load Control

Contents

This section contains information on the following topics:

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 23: LLC, system control levels (hierarchy and overlap of operative levels) on page 328 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 telephone is to receive a dial tone. Sets that are blocked during an LLC level upgrade do not receive a dial tone.

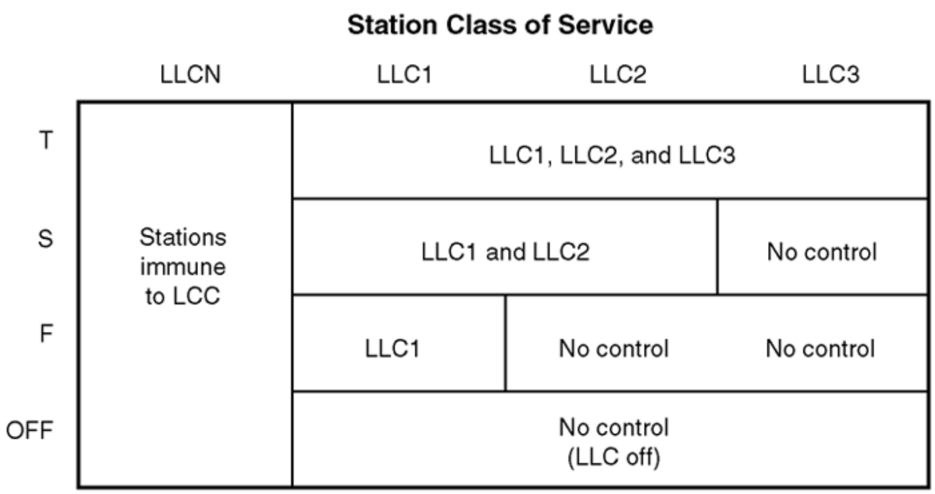


Figure 23: LLC, system control levels (hierarchy and overlap of operative levels)

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. Table 99: LD 10 on page 330

Add or change Line Load Control for analog (500/2500-type) telephones.

2. Table 100: LD 11 on page 330

   Add or change Line Load Control for Meridian 1 proprietary telephones.

3. Table 101: LD 2 on page 330

   Set Line Load Control levels.

Table 99: LD 10

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	500	Telephone type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	(LLCN) LLC1 LLC2 LLC3	LLC not enabled (default) LLC class 1 LLC class 2 LLC class 3

Table 100: LD 11

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	(LLCN) LLC1 LLC2 LLC3	LLC not enabled (default) LLC class 1 LLC class 2 LLC class 3

Table 101: LD 2

Prompt	Response	Description
SCTL	x aaa	Set blocking probability

Prompt	Response	Description
		x = F (LLC, level F) x = S (LLC, level S) x = T (LLC, level T) aaa = 0-100
SLLC	x	Activate LLC at level x x = F (LLC, level F) x = S (LLC, level S) x = 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.

Chapter 43: Line Lockout

Contents

This section contains information on the following topics:

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 telephone in Line Lockout state, busy tone is 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) is 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

Table 102: LD 15 - Implement Flexible Line Lockout for a customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	INT	Intercept treatment options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
ICI	0-19 LCT	Assign a Flexible Line Lockout Incoming Call Indicator (ICI) key to attendant consoles.
- LLT	 (OVF) OFA ATN	Line Lockout treatment. Overflow tone, then lockout. Overflow tone, then attendant intercept. Attendant intercept
- DLT	 (OVF) OFA ATN	Line lockout treatment for Direct Inward System Access (DISA) calls. Overflow tone, then lockout. Overflow tone, then attendant intercept. 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.

Chapter 44: Listed Directory Numbers

Contents

This section contains information on the following topics:

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

Table 103: LD 15 - Assign Listed Directory Numbers for each customer.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	LDN	Departmental Listed Directory Numbers
CUST		Customer number

Prompt	Response	Description
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
- 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.

Chapter 45: Listed Directory Numbers, Network Wide

Contents

This section contains information on the following topics:

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 is 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 is 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 are 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 is 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 is presented on the LDN ICI.

Feature packaging

Because 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. Table 104: LD 15 on page 346

   Activate Network Wide LDN in CDB.

2. Table 105: LD 15 on page 347

   Add or change LDN keys.

3. Table 106: LD 93 on page 347

   Add or change LDN keys in CPG.

Table 104: LD 15

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. 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.
...		
- 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.

Prompt	Response	Description
- LDA5	xx xx... ALL	Attendant consoles associated with LDN5.
- OPT	NLDN () XLDN	Enable network wide LDN. Exclude LDN.

Table 105: LD 15

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.

Table 106: LD 93

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.

The feature option in the originating 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.

Chapter 46: Lockout, DID Second Degree Busy, and MFE Signaling Treatments

Contents

This section contains information on the following topics:

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 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 in Second Degree Busy (that is, a telephone 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; 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

Define an option for DID calls to a second degree busy telephone, and define a new intercept treatment for calls in a lockout state.

Table 107: LD 15

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 telephone are disconnected. If denied, calls forwarded to a busy telephone follow the telephone 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.

Chapter 47: LOGIVOX Telephone

Contents

This section contains information on the following topics:

Feature description

The LOGIVOX is a digital telephone 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 telephone.

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 telephone has display denied Class of Service. If the telephone has LOGIVOX denied Class of Service, each digit is shown twice.

On-hook Dialing

Because of the firmware on the LOGIVOX telephone, 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

Table 108: 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.

Chapter 48: Loop Start Supervisory Trunks

Contents

This section contains information on the following topics:

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 (see 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

Because Loop Start Supervisory trunks do not provide disconnect supervision on incoming calls, Nortel does not recommend 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 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 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. Table 109: LD 14 on page 360

Create or modify trunk data blocks on a per trunk basis.

2. Table 110: LD 16 on page 360

Table 109: LD 14

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.

Table 110: LD 16

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.

Chapter 49: Loop Start Supervisory Trunks (Incoming Calls)

Contents

This section contains information on the following topics:

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

Table 111: 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.

Chapter 50: Loopback on Public Exchange or Central Office Trunks

Contents

This section contains information on the following topics:

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.

Chapter 51: M3900 Full Icon Support

This section contains information about the following topics:

Feature description

Using distinct icons and flashing cadences, the M3900 Full Icon Support feature informs users of various call states. The icons appear on the LCD next to the Directory Number (DN) keys on the following telephones:

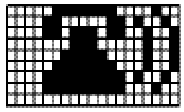
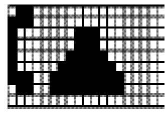
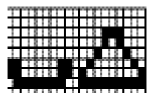
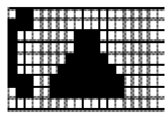
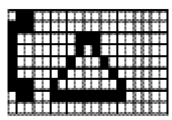
- M3903 series digital telephones (Phase II and Phase III)
- M3904 series digital telephones (Phase II and Phase III)
- M3905 series digital telephones (Phase III)

The icons also appear on the Key-Based Accessory module and the Display- Based Accessory module.

The Full Icon Support feature provides icons for the following functions:

- I-Ringing: The I-Ringing icon appears on the ringing DN of the called telephone.
- I-Active: Telephones in an active call state display the I-Active icon.
- U-Active: The U-Active icon appears on the Multiple Appearance Directory Number (MADN) of a telephone when another telephone sharing the MADN is in the active call state.
- I-Hold: The I-Hold icon appears on the DN of a telephone with a call on hold.
- U-Hold: The U-Hold icon appears on the MADN of the telephone when another telephone sharing the MADN has a call in the hold state.

With Full Icon Support enabled, the Ringing, I-Hold, U-Hold, and Active DN keys display the icons in the following table.

Call/Feature state	DN key icon	Cadence
Ringing		Flash
I-Hold		Wink
U-Hold		Flicker
I-Active		On
U-Active		On

Operating parameters

The M3900 Full Icon Support feature requires a minimum of Release 9 of the Key-Based Accessory (KBA) module.

Feature interactions

No feature interactions are associated with this feature.

Feature packaging

The M3900 Full Icon Support feature requires the following packages:

- M3900 Full Icon Support (ICON_PACKAGE) package 397
- Digital Sets (DSET) package 88

Feature implementation

To enable M3900 Full Icon Support, use LD 17.

Table 112: LD 17 - Enable M3900 Full Icon Support

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	PARAM	System parameters
.....		
ICON	(NO) YES	Enable the M3900 Full Icon Support Feature NO = Disable the M3900 Full Icon Support Feature

Feature operation

No specific operating procedures are required to use this feature.

Chapter 52: M3900 Set-to-Set Messaging

This section contains information about the following topics:

Feature description

Use the Set-to-Set Messaging feature to send text messages from one M3900 telephone (called party) to another M3900 telephone (calling party). On receiving a call, an enabled telephone sends a single text message, as specified from the Applications menu.

The following telephones support Set-to-Set Messaging:

- M3903
- M3904
- M3905

During active Set-to-Set Messaging, the caller receives both an audible tone and the sent message. After that, the caller receives a ringback tone, and the call switches to voice messaging. If the called telephone is busy, the calling party receives a call-waiting tone.

Use the Applications menu to activate Set-to-Set Messaging and select which message you want to send. To access the Applications menu, press the Applications key.

Table 4 shows some sample Set-to-Set Messages.

Table 113: Examples of messages text

OUT TO LUNCH
BACK TO WORK: 4 Dec 02
BACK TO OFFICE: Jan 03
WILL REPLY AFTER 1 PM

BACK @ 4:00 PM
NOT IN TODAY
RETURN SOON - 8:10 PM
GONE FOR THE DAY

Operating parameters

Set-to-Set Messaging accepts a maximum of 24 characters per configured message (equivalent to one line on the telephone display).

Using the Applications menu, you can configure up to 10 text messages. However, M3900 telephones support only one Set-to-Set message at a time. To activate Set-to-Set Messaging, you must first define at least one message.

Password protection (if active) also applies to Set-to-Set Messaging.

Feature interactions

Multiple Appearance Redirection Prime/Multiple Appearance Directory Number

If the Multiple Appearance Redirection Prime (MARP) feature is active, then MARP determines which DNs receive the Set-to-Set Message. If MARP is inactive, then Multiple Appearance Directory Number (MADN) determines which DNs configured on the telephone receive the Set-to-Set Message.

Feature packaging

M3900 Set-to-Set Messaging requires the Set-to-Set Messaging package 380.

Feature implementation

Task summary list

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

- Table 114: LD 11 on page 373

  Configure Class of Service for M3900 Set-to-Set Messaging
- Table 115: LD 15 on page 373

  Modify Set-to-Set Messaging

Table 114: LD 11

Prompt	Respone	Description
REQ	NEW CHG	Add new data Change existing data
TYPE	a..a	Type of telephone 3903 = M3903 3904 = M3904 3905 = M3905 Only M3903, M3904, and M3905 support Set-to-Set Messaging.
...		
CLS	(STSD) STSA	(Deny) Allow Set-to-Set Messaging

Table 115: LD 15

Prompt	Respone	Description
REQ	NEW CHG	Add new data Change existing datablock information
TYPE:	FTR	Features and options
CUST	0-99 0-31	Customer number Range for Large System and CS1000E system Range for Small System and Media Gateway 1000B.
...		
STS_MSG	(NO) YES	Modify Set-to-Set Messaging
MSG 01	<CR> <text string>	Keeps current message Input the new message to be displayed (up to 24 characters)

Prompt	Respone	Description
...		
MSG 10	<CR> <text string>	Keeps current message Input the new message to be displayed (up to 24 characters)

Feature operation

For more information about the operation of Set-to-Set Messaging, see the *Meridian Digital Telephones: M3901, M3902, M3903, M3904 User Guide*.

Chapter 53: M3900 (Single Site) Virtual Office

Contents

This section contains information on the following topics:

Feature description

See IP Network-wide Virtual Office on page 243 andIncremental Software Management on page 764 for more information on related features and functionality.

Terms

Host Terminal—A physical M3903 or M3904 telephone at which a user logs in to begin a Virtual Office session.

Virtual Office worker—The user who logs into the physical telephone and uses their own personal Virtual Office configuration at the Host Terminal.

Virtual Set— The user personal, non-physical-telephone configuration programmed on a Phantom TN.

Description

The M3900 (Single Site) Virtual Office feature allows a user to log in to a designated M3903 or M3904 telephone (Host Terminal) and use an individual telephone configuration (Virtual Set configuration) at that telephone. Calls to the DN assigned to the Virtual Set (the user primary DN) are routed to the Virtual Office Host Terminal where the Virtual Office worker is logged in. The Host Terminal is the physical telephone that a user logs in to as a Virtual Office worker.

This feature maximizes the use of office space and desktop equipment by offering functionality referred to as "Hotelling" or "Hot-desk". This feature allows office space to be set up with designated, host telephones for the use of visiting telecommuters who can log in using a Flexible Feature Code and their individual DN.

This capability is useful for telecommuters, visitors, and workers who are frequently out of the office. The visitor can log in at any one of the designated telephones set aside for this purpose.

M3903 and M3904 telephones can be configured as Host Terminals.

A Virtual Office worker is required to log in to a Host Terminal that matches their Virtual Set telephone type. For example, when the individual configuration (the Virtual Set) of a Virtual Office worker is configured as an M3904, the logon process is blocked, if they attempt to log in to an M3903 Host Terminal.

The Virtual Set is a set of features configured for a user and defined on a Phantom loop. There is no permanent physical telephone associated with a Virtual Set.

The M3900 (Single Site) Virtual Office feature operates on stand-alone Meridian 1 and CS 1000 systems only.

The Virtual Office worker is identified by a primary DN, which cannot be used as the primary DN for any other telephone, virtual or physical, in the system.

Use the Station Control Password (SCPW, configured in LD 11), to validate the logon.

To log in using the M3900 (Single Site) Virtual Office feature, the TN associated with the Host Terminal must be configured with the Virtual Office Login Allowed (VOLA) Class of Service (CLS). The TN associated with the User ID for the logon must be configured with the CLS VOUA (Virtual Office User Allowed). For more information on VOLA and VOUA, see LD 11 and LD 81 in *Software Input Output - Administration, NN43001-611*.

Nortel recommends that the Host Terminal have at least internal call and emergency call (911 in North America) capability.

Clearing of the Directory Services Password

With M3900 Phase III, the system can clear the Directory Services password when a Virtual Office worker logs in or out of an M3903 or M3904 Host telephone, if Erase List is allowed. The system administrator configures this functionality by defining the Class of Service as Erase List Allowed (ELA) in LD 11 for the M3903 or M3904 Virtual Set.

This Clearing of Password functionality allows multiple virtual workers, using the same Host telephone, to have access to password-protected features if one of the users sets the password and does not turn it off when they log out.

Clearing of the Callers List and Redial List

The contents of the Redial List, Personal Directory, and Call Log are stored on the M3900 telephone itself between login sessions.

With M3900 Phase III, the system can clear the Redial and Callers lists when a Virtual Office worker logs in or out of an M3903 or M3904 Host telephone. The system administrator configures this functionality by defining the Class of Service as Erase List Allowed (ELA) in LD 11 for the M3903 or M3904 Virtual Set. When the ELA Class of Service is defined, the Callers List and Redial List are automatically cleared when the virtual worker logs in or out. If ELA is not defined, other workers using the same Host telephone can view the Callers and Redial Lists of the previous users.

Automatic Logout for Virtual Office

M3900 Phase III introduced automatic logout for Virtual Office workers. If a Virtual Office worker, who is already logged on to telephone A, tries to log on to telephone B, the system automatically logs the Virtual Office worker off at telephone A and logs them on to telephone B (provided that the Virtual Office worker enters the correct login password). The system administrator enables this functionality in LD 15 at the Virtual Office Automatic Logout (VO_ALO) prompt.

The system administrator can also define a time at which all Virtual Sets are automatically logged out. The system administrator configures the automatic logout time at the Virtual Office Automatic Logout Time (VO_ALOHR) prompt in LD 15.

If the telephone is busy at the automatic logout time (for example, if the Virtual Office worker is using Corporate Directory or Set-to-Set Messaging), logout occurs when the telephone becomes idle.

If a user logs in to a Host telephone after automatic logout has occurred, the telephone does not automatically log out a second time.

Speed Call for Virtual Office

With M3900 Phase III, M3900 telephones support Speed Call (SCU/SCC) and System Speed Call (SSU only) on Virtual Set TNs.

Operating parameters

Only one active session per user login ID is allowed at one time in the system.

The Virtual Set Primary DN cannot be a Primary DN on another terminal. The Primary DN of Virtual Set A can be the secondary DN of another Virtual Set. If both Virtual Set users are logged in, a call to user A Primary DN can be answered by user B Secondary DN.

If Virtual Office worker A logs out, Virtual Office worker B logs in, and a user calls the Primary DN of Virtual Set A, the scenarios are as follows:

- If Virtual Office worker A has Call Forward configured before logout, the call is forwarded.
- If Virtual Office worker A does not have Call Forward configured, but has the default Call Forward (DCFW) configured, the call is forwarded to the DCFW DN. (The DN can be a voice mail DN.)
- If neither of the above two scenarios apply, the caller receives overflow tone.

The Virtual Set recognizes all system configurations related to the user.

Feature interactions

Speed Call

With M3900 Phase III, M3900 telephones support Speed Call (SCU/SCC) and System Speed Call (SSU only) on Virtual Set TNs.

Feature packaging

M3900 (Single Site) Virtual Office requires the following packages:

- Virtual Office (VO) package 382
- Virtual Office Enhancement (VOE) package 387

Feature implementation

Task summary list

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

1. Table 116: LD 97 on page 379

   Configure a Phantom loop for Virtual Set.
2. Table 117: LD 57 on page 379

   Configure Virtual Office Flexible Feature Codes.
3. Table 118: LD 11 on page 380

   Configure the Virtual Set with Erase List Allowed.
4. Table 119: LD 81 on page 380

   Print a list or count of Virtual Office terminals.
5. Table 120: LD 20 on page 381

   Print Terminal Number Block (TNB) data for Virtual Sets and Host Terminals.

Table 116: LD 97

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	SUPL	Superloops.
SUPL	Naaa	Superloop designation, where N designates the superloop as a Phantom loop, and aaaa = superloop number.
...	...	...

Table 117: LD 57

Prompt	Response	Description
REQ:	NEW	Add new data block information.
	CHG	Change data block information.
	OUT	Remove data block information.
	END	Exit data block overlay program.

Prompt	Response	Description
TYPE	FFC	Flexible Feature Codes data block.
...	...	
CODE	VTLN	FFC for Virtual Set login.
	ALL	Every FFC is prompted.
	<CR>	No further prompt; returns to REQ.
VTLN	xxxx	Virtual Set login code.
	<CR>	Returns to "CODE"
CODE	VTLF	FFC type for Virtual Set logout.
	ALL	Every FFC is prompted.
	<CR>	No further prompt; returns to REQ.
VTLF	xxxx	Virtual Set logout code.
	<CR>	Returns to "CODE" A Phantom TN cannot be moved or copied.

Table 118: LD 11

REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	(ELD) ELA	Erase Lists (Denied) Allowed.

Table 119: LD 81

Prompt	Response	Description
REQ	LST	Print a list of telephones.
	CNT	Print a count of telephones.
CUST	xx	Customer number, as defined in LD 15.
...	...	...
FEAT	aaa	Designate a feature mnemonic.
	3900	Print M3900-type telephones, including Virtual Sets and Host Terminals.

Prompt	Response	Description
	DCFW	Print default call forward for Phantom TNs, including Virtual Sets.
...	...	...

Table 120: LD 20

Prompt	Response	Description
REQ:	PRT	Print data block for the requested terminal type(s).
	LTN	List Terminal Numbers of the requested terminal type(s).
TYPE:	a...a TNB	Telephone type, where a...a = 3903V (M3903 Virtual Set), 3904V (M3904 Virtual Set), 3903H (M3903 Host Terminal), or 3904H (M3904 Host Terminal). The only telephone types of the M3900 Series that can be configured as a Virtual Set or Host Terminal are the M3903 and M3904. The Print TNB and List TNB requests always show the logged-off TNB data. In logged-in state, an indication of the logged-in TN ("HOST TN" or "VIRTUAL TN") is added.

Feature operation

For more information on the operation of this feature, see the *Meridian Digital Telephones: M3901, M3902, M3903, M3904 User Guide*.

Chapter 54: Make Set Busy and Voice Call Override

Contents

This section contains information on the following topics:

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 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.

Operating parameters

A Voice Call key on a Meridian 1 proprietary telephone can only be programmed to a single appearance DN.

The telephone being voice called must be equipped with a speaker.

Feature interactions

Do Not Disturb

Voice calls are not allowed on a telephone 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

Table 121: 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		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
...		

Prompt	Response	Description
- 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 telephone with MSB active:

In this example, telephone A is a Meridian 1 proprietary telephone with a VCC key programmed with the DN of a single appearance key on telephone B.

Telephone B is a Meridian 1 proprietary telephone with a single appearance DN key. Telephone 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 VCC key lamp is lit and A receives ringback tone. B receives a two-second burst of ring tone. B terminating DN key lamp flashes.
3. After two seconds:

   Telephone A has a one-way voice path to telephone B. B DN key lamp is lit. Ring tone to B stops. Ringback tone to A stops. B Make Set Busy lamp remains lit.
4. If B goes off-hook, A and B are connected in a normal two-way conversation.

Chapter 55: Make Set Busy

Contents

This section contains information on the following topics:

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 telephone that has the Make Set Busy feature is idle, the DN is blocked and if the DN is busy, busy tone is 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 telephone.

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

For more information about MSB operations, see *Automatic Call Distribution Fundamentals, NN43001-551*.

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 telephone 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.

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 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.

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 telephone is considered busy while MSB is active. A CCBS request is registered against a busy telephone, but only advances when the MSB feature is deactivated and the telephone 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.

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

Table 122: LD 11 - Add or change MSB for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
KEY	xx MSB	Add an MSB key. 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.

Chapter 56: Make Set Busy Improvement

Contents

This section contains information on the following topics:

Feature description

This feature is designed for a boss/secretary environment. The same Directory Number (DN) appears on more than one telephone, and is defined as ringing on the secretary telephone and non-ringing on the boss telephone.

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 telephone, 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-type) telephones. However, the ringing appearances of the DN can be an analog (500/2500-type) telephone but not for a private line.

Feature interactions

Short buzz for Digital sets

If the MSB key is activated on a telephone, and there is an incoming call to another SCN/MCN DN key on the same telephone, 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) telephones 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. Table 123: LD 11 on page 397

   Activate the MSBI feature and define Primary DN telephone (boss) with non-ringing DN key.
2. Table 124: LD 11 on page 398

   Define another telephone (secretary) with ringing DN key and MSB key.

Table 123: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
...	...	
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.
...	...	

Table 124: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
...	...	
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.

Chapter 57: Malicious Call Trace

Contents

This section contains information on the following topics:

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) and Meridian digital telephones (Meridian 1 proprietary telephones), or by key access from 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.

  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 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 telephone 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.

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 telephone 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 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.

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.

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. Table 125: LD 10 on page 407

   Enable MCT on an analog (500/2500-type) telephone.
2. Table 126: LD 11 on page 407

   Enable MCT on a proprietary telephone.
3. Table 127: LD 17 on page 408

   Allow printing of the MCT record on a dedicated MCT TTY port.
4. Table 128: LD 16 on page 408

   Set up the recorder route.
5. Table 129: LD 14 on page 408

   Set up the recorder trunk.
6. Table 130: LD 15 on page 409

   Set up the recorder and alarm options.
7. Table 131: LD 16 on page 410

   Set up the alarm for external calls.
8. Table 132: LD 57 on page 410

   Define the MCT FFC.
9. Table 133: LD 16 on page 410

Configure the call trace string.

10. Table 134: LD 14 on page 411

    Enable Firmware timing for trunk hook flash (if available).

11. Table 135: LD 73 on page 412

    Define the DTI2 flash time range.

12. Table 136: LD 16 on page 412

    Set up MCTM timer and tandem delay (2 Mbps PRI for AXE-10 Australia only).

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.

Table 125: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	500	Analog (500/2500-type) telephone data block.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	MCTA, MCTD	Malicious Call Trace is allowed if Class of Service is MCTA.

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.

Table 126: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Table 127: LD 17

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.

Table 128: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route data block.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
TKTP	RCD	Recorder trunk data block.
ACOD	xxxx	Recorder route access code.

Table 129: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RCD	Recorder trunk.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.

Table 130: LD 15

Prompt	Response	Description
REQ:	CHG PRT END	Change, print, or end.
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
...		
- 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.

Table 131: LD 16

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.

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.

Table 132: LD 57

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.

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 must be defined.

Table 133: LD 16

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.

Prompt	Response	Description
		• 60 to 89 ms = Digit 1 is sent • 90 ms = Hard coded for XFCOT hook flash • 91 to 255 ms = Digit 1 is sent • 256 to 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 to 127 msec is done by sending digit 1. The range of 128 to 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 to 9, *, and #.
MCDT	(0)-4	Digit string delay is in seconds, in increments of one second.

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.

Table 134: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DID COT	Trunk type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
XTRK	EXUT XCOT	Card type

Prompt	Response	Description
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		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.

Table 135: LD 73

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.

Table 136: LD 16

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:

Press Call Trace. You remain connected to the call.

Chapter 58: Malicious Call Trace DN/TN Print

Contents

This section contains information on the following topics:

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 on page 399 feature in this document.

Feature interactions

See the Malicious Call Trace on page 399 feature in this document.

Feature packaging

Malicious Call Trace (MCT) package 107.

Feature implementation

Table 137: 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#xxxxxxxxxxxxxxxx

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#xxxxxxxxxxxxxxxx++
2. Terminator<*>DNxxxxxxx++++++++++++++

or

1. Originator<*>DNxxxxxxx++++++++++++++
2. Terminator<*>CLID#xxxxxxxxxxxxxxxx++

Chapter 59: Malicious Call Trace Idle

Contents

This section contains information on the following topics:

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 is 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. Table 138: LD 10 on page 421

   Enable MCT on an Analog (500/2500-type) telephone.
2. Table 139: LD 11 on page 422

   Enable MCT on a proprietary telephone.
3. Table 140: LD 15 on page 422

   Enable the MCT signal.
4. Table 141: LD 94 on page 422

   Create or modify the MFC tables.
5. Table 142: LD 16 on page 422

   Create or modify data for each DID trunk route data block to allow or deny MFC Signaling option.
6. Table 143: LD 14 on page 423

   - Create or modify data for each DID trunk data block to allow or deny MFC Signaling option.

Table 138: LD 10

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.

Table 139: LD 11

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.

Table 140: LD 15

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.

Table 141: LD 94

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.

Table 142: LD 16

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.

Table 143: LD 14

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.

Chapter 60: Malicious Call Trace on Direct Inward Dialing

Contents

This section contains information on the following topics:

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. Table 144: LD 10 on page 426

   - Enable MCT on an analog (500/2500-type) telephone data block.

2. Table 145: LD 11 on page 427

   Enable MCT on a Meridian 1 proprietary telephone data block.

3. Table 146: LD 15 on page 427

   Enable MCT signal

4. Table 147: LD 94 on page 427

   Create or modify the MFC tables.

Table 144: LD 10

Prompt	Response	Description
...		
CLS		Class of Service.
	MCTA	Malicious Call Trace allowed. When MCTA is assigned, the station must also have XFA defined.

Prompt	Response	Description
	(XFD) XFA	Call Transfer (denied) allowed.

Table 145: LD 11

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.

Table 146: LD 15

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.

Table 147: LD 94

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.

Chapter 61: Manual Line Service

Contents

This section contains information on the following topics:

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

Table 148: LD 10 - Define Class of Service for Manual Line telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Chapter 62: Manual Service Recall to Attendant

Contents

This section contains information on the following topics:

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

Table 149: 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.

Chapter 63: Manual Signaling (Buzz)

Contents

This section contains information on the following topics:

Feature description

Manual Signaling (Buzz) permits a Meridian 1 proprietary telephone user to sound a buzz tone at a specific telephone.

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 telephone.

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

Table 150: LD 11 - Add Manual Signaling (Buzz) key for Meridian 1 proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Chapter 64: Manual Trunk Service

Contents

This section contains information on the following topics:

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. Table 151: LD 16 on page 440

   Add or change an incoming manual trunk route.
2. Table 152: LD 14 on page 441

   Add or change an incoming manual trunk.
3. Table 153: LD 16 on page 441

   Add or change an outgoing manual trunk route.
4. Table 154: LD 14 on page 442

   Add or change an outgoing manual trunk.

Table 151: LD 16

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		Route number

Prompt	Response	Description
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
TKTP	TIE	Incoming manual trunks (must be TIE trunks).
ICOG	ICT	Incoming route.
ACOD	xxxx . . x	Trunk route access code.

Table 152: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	TIE	TIE trunks are required for manual incoming trunks.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.
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.

Table 153: LD 16

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		Route number

Prompt	Response	Description
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
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.

Table 154: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaa	Outgoing trunk type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.
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.

Chapter 65: Meridian 1 Attendant Console Enhancements

Contents

This section contains information on the following topics:

Feature description

The Meridian 1 Attendant Console Enhancements (MACE) feature expands existing 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 to 31 digits in length and can be either internal or external to the system.

The Autoline key 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 24: Attendant console with four Autoline keys configured on page 444 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 DN.

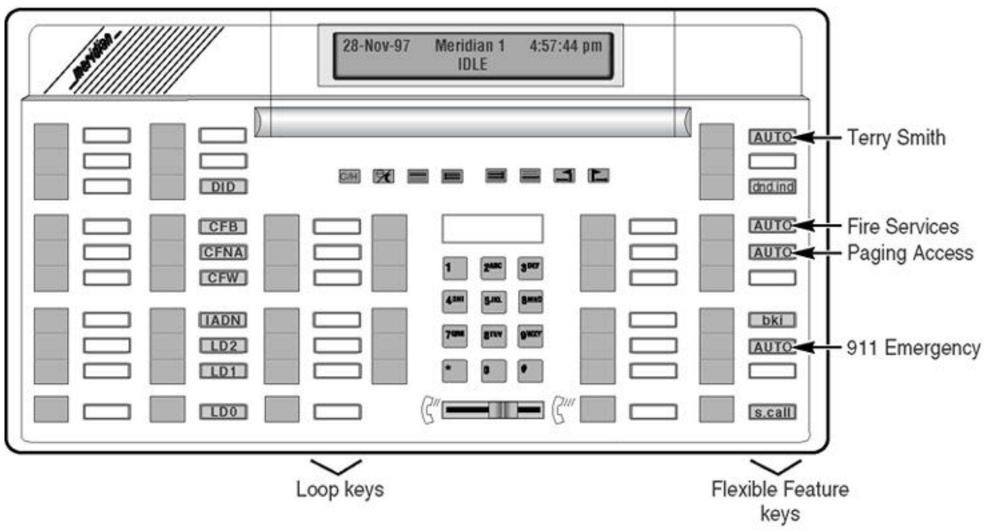


Figure 24: Attendant console with four Autoline keys configured

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 25: Attendant display when the Autoline functionality is in progress on page 445 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 DN. In this case, once the Autoline key is pressed, the attendant display is as shown in Figure 25: Attendant display when the Autoline functionality is in progress on page 445.

Figure 25: Attendant display when the Autoline functionality is in progress

Figure 26: Attendant display when the attendant places an Autoline call to an external Autoline DN on page 445 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 26: 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 27: Attendant display when the attendant extends a call to the Autoline DN on page 445 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 DN. Once the Autoline key is pressed, the attendant display is as shown in Figure 27: Attendant display when the attendant extends a call to the Autoline DN on page 445. When the attendant presses the Release key, the display is cleared.

Figure 27: Attendant display when the attendant extends a call to the Autoline DN

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 telephone.

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 telephone 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 28: Attendant console with an IADN ICI key configured on page 446 illustrates an attendant console with an IADN ICI key configured on Key Strip 2. This key strip holds the ICI keys.

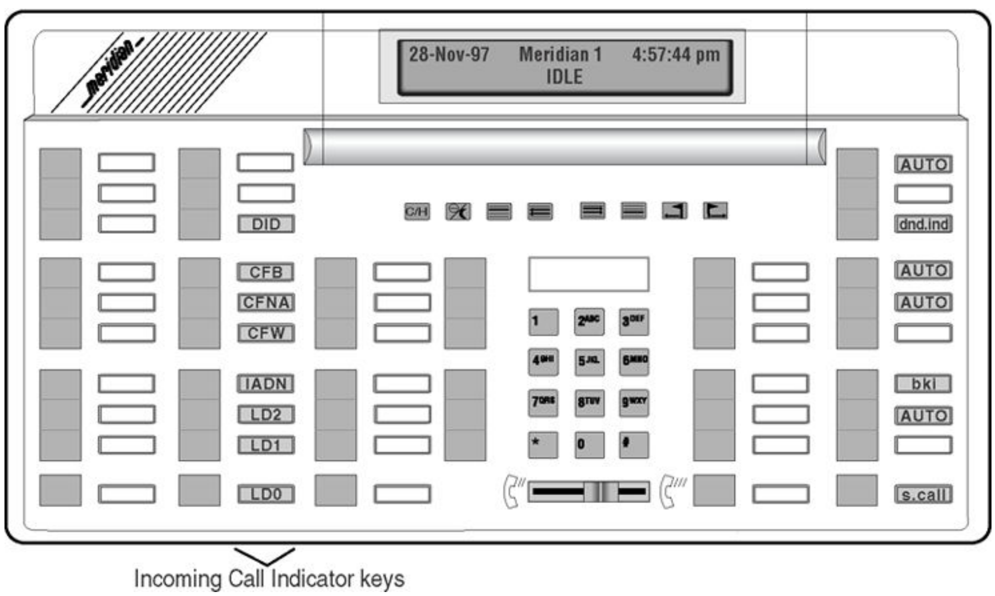


Figure 28: Attendant console with an IADN ICI key configured

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.

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 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 telephone 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 telephone 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 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 29: An example of attendant console DLDN groupings on page 450 provides an example of attendant console DLDN groupings. These groups are assigned using the LDA prompt in LD 15. In Figure 29: An example of attendant console DLDN groupings on page 450, 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.

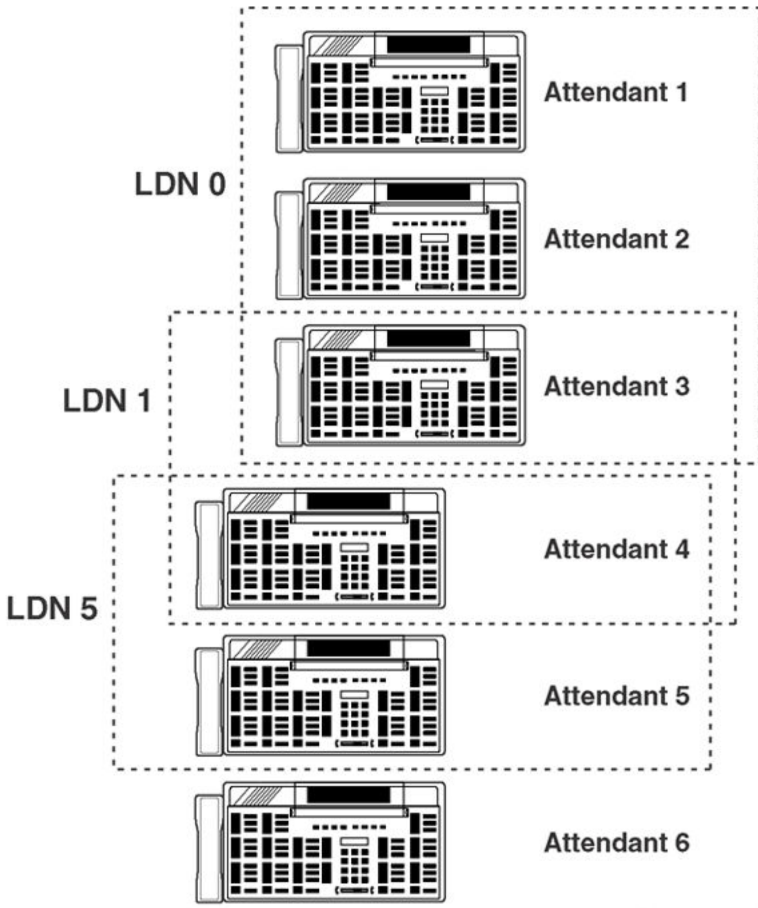


Figure 29: An example of attendant console DLDN groupings

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 30: Attendant console with three LDN ICI keys configured on page 451 shows an example of an 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.

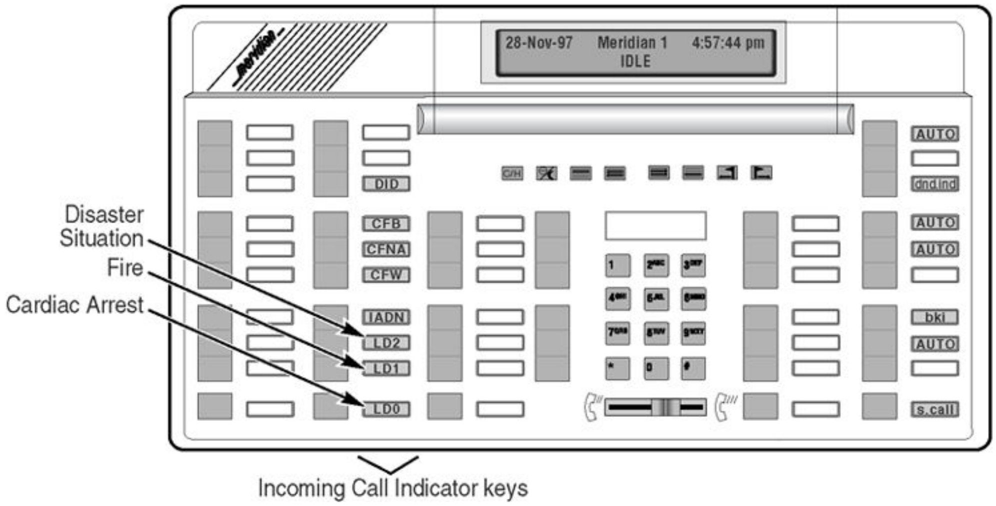


Figure 30: Attendant console with three LDN ICI keys configured

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, because the call is already presented on the Loop key.

Referring to Figure 29: An example of attendant console DLDN groupings on page 450, consider the following example:

1. Party 1, an internal telephone 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 29: An example of attendant console DLDN groupings on page 450, consider the following example:

1. Party 1 (an internal telephone 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 29: An example of attendant console DLDN groupings on page 450, consider the following example regarding an active attendant console:

1. Party 1 (an internal telephone 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 29: An example of attendant console DLDN groupings on page 450, 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. Because 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 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 are 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) telephone 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, telephone recalls do not take priority in the attendant queue.

When an analog (500/2500-type) telephone without CLS = XFA/TSA performs a switch hook flash or when a Meridian 1 proprietary telephone 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) telephone with CLS = XFA/TSA performs a switch hook flash and then dials an IADN, the call is treated as a regular telephone 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 telephone 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 telephone belonging to one tenant can call an attendant belonging to another tenant by dialing the attendant 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 telephone is in Permanent Hold and dials an IADN, the telephone 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. Table 155: LD 15 on page 463

   Configure Priority Buzzing and an Individual Attendant Directory Number (IADN) Incoming Call Indicator (ICI) key for a digital attendant console.
2. Table 156: LD 15 on page 464

   Configure Departmental Listed Directory Number (DLDN) and Priority Buzzing for Attendant Emergency Code calls.
3. Table 157: LD 12 on page 464

   Configure an Autoline DN for an attendant console.
4. Table 158: LD 12 on page 465

   Configure an Individual Attendant Directory Number (IADN) for a digital attendant console.

Table 155: LD 15

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	2250	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.

Prompt	Response	Description
...		
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).

Table 156: LD 15

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.

Table 157: LD 12

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.

Prompt	Response	Description
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
...		
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.

Table 158: LD 12

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
ANUM	1 - 63	Attendant Number.
...		
IADN	xxxx	Individual Attendant DN for this attendant console. The Individual Attendant DN can be 1 to 4 digits in length or 1 to 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 Autoline 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 29: An example of attendant console DLDN groupings on page 450:

1. Party 1 (an internal telephone 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.

Chapter 66: Meridian 1 Initialization Prevention and Recovery

Contents

This section contains information on the following topics:

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.

Chapter 67: Meridian 911

Contents

This section contains information on the following topics:

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 familiarity with an area, or the caller 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 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
- the billing address associated with that telephone number

To protect a caller 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.

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, see the "10/20 Digit ANI on 911 Calls" feature description in *Features and Services Reference (NN43001-106).*

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 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 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.

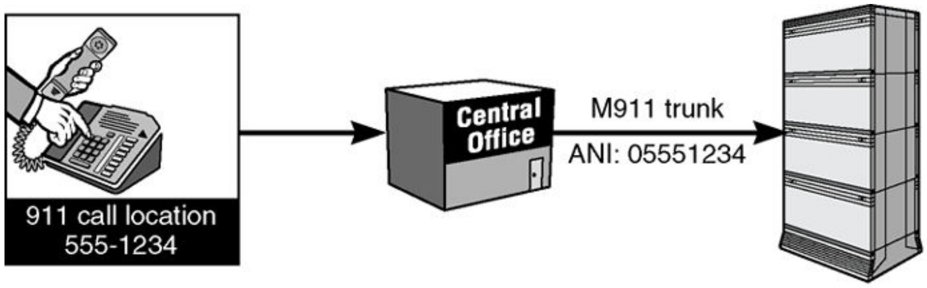


Figure 31: Routing the call, along with the ANI digits, to the system

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.

M911 Networked Operation

For information about the M911 Networked Operation feature, see *Emergency Services Access Fundamentals, NN43001-613*and *Automatic Call Distribution Fundamentals, NN43001-551*.

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 telephone 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 limited 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 is 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 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 are 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 because 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 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 is 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, because 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 is linked into the default destination ACD DN 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 because the Route To, Disconnect, and Busy treatments remove CDN control from the call, ICCM messages are 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 limited to ACD DNs only. If the routing destination is not an ACD DN, the call is routed to the CDN 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 telephone 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 is 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 limitation is lifted for 911 calls only.

The CWNT telephone 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, is 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 telephone regardless of the calling party 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

Because 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 telephone).

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, because 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. Because 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 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.

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

Because there is no speech path between the ACD agent and the caller, the supervisor observe feature is 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. Table 159: LD 10 on page 489

   Configure a Terminal Number for an analog (500/2500-type) telephone with a Class of Service of CWNA (Call Waiting Notification Allowed).
2. Table 160: LD 23 on page 490

   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. Table 161: LD 23 on page 490

   Configure CDNs. The ceiling value is irrelevant for 911 calls terminating at the CDN, but is applied to non-911 type calls. When the ceiling value is exceeded, new non-911 calls will receive busy tone.
4. Table 162: LD 16 on page 491

   Configure an M911 route.
5. Table 163: LD 16 on page 491

   Create a Numbering Plan or Information Digit (NPID) Table.
6. Table 164: LD 14 on page 492

   Configure 911 trunks.
7. Table 165: LD 16 on page 493

   Configure Call Detail Recording (CDR).
8. Table 166: LD 17 on page 493

   Configure the insertion of ANI digits into the CDR record.
9. Table 167: LD 10 on page 493

   Configure non-ACD sets (analog [500/2500-type] telephones).
10. Table 168: LD 11 on page 494

    Configure non-ACD sets (Meridian 1 proprietary telephones).
11. Table 169: LD 11 on page 494

    Configure Meridian 1 proprietary telephones to function as ACD sets.
12. Table 170: LD 16 on page 495

    Enable M911 Call Abandon.
13. Table 171: LD 56 on page 496

    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.

Table 159: LD 10

Prompt	Response	Description
REQ:	NEW	New.
TYPE:	500	Type of telephone.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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).

Table 160: LD 23

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.

Table 161: LD 23

Prompt	Response	Description
REQ	NEW	New.

Prompt	Response	Description
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).

Table 162: LD 16

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		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
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.

Table 163: LD 16

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.

Prompt	Response	Description
TRMT	(NONE) NPA FAIL TEST	Numbering Plan Digit or Information Digit treatment.
- NPA	nnn	Numbering Plan Area. Prompted only if TRMT = NPA.

Table 164: LD 14

Prompt	Response	Description
REQ	NEW	New.
TYPE	DID	Meridian 911 trunks must be DID.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
...		
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		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.
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).

Table 165: LD 16

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
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.

Table 166: LD 17

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters
...		
- CLID	YES	Calling Line ID (ANI for M911) in CDR.

Table 167: LD 10

Prompt	Response	Description
REQ:	NEW	Add a telephone.
TYPE:	500	Type of telephone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Prompt	Response	Description
DN	nn...n	Directory Number.
...		
IAPG	2	ISDN/AP status message group.
...		
CLS	USMA	Unsolicited Status Allowed Class of Service. M911 position.

Table 168: LD 11

Prompt	Response	Description
REQ:	NEW	Add a telephone.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Table 169: LD 11

Prompt	Response	Description
REQ:	NEW	Add a telephone.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Table 170: LD 16

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		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
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.

Prompt	Response	Description
M911_TONE	(YES) NO	Optional call abandon tone, where: YES = tone given on answer, and NO = silence given on answer.

Table 171: LD 56

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 receiver telephone.

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 receiver). Once the call receiver 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 receiver must hang up and dial the ANI that is shown on the terminal display if call back is required.

Upon answer, the telephone display is updated with the 911 call receiver ANI and the trunk group name if the Call Party Name Display feature is used. Because the call has been abandoned, the telephone display flags the abandoned call by appending "ABAND" to the ANI.

Figure 32: Display for an NPD call on page 497 shows what is displayed on a telephone 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 telephone display; the ANI appears on the second line.

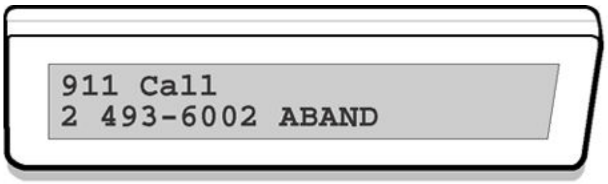


Figure 32: Display for an NPD call

Figure 33: Display for an NPA call on page 497 shows a telephone 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 telephone display. The ANI appears on the second line.

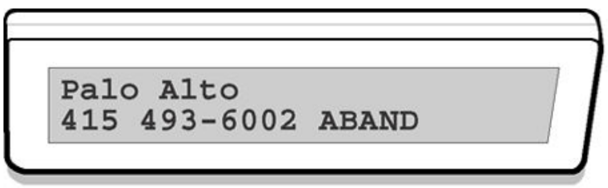


Figure 33: Display for an NPA call

Chapter 68: Meridian Hospitality Voice Services

Contents

This section contains information on the following topics:

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 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 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 and Meridian Modular softkey menus

M2317 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

See *Hospitality Features Fundamentals, NN43001-553*

Feature operation

See *Hospitality Features Fundamentals, NN43001-553*.

Chapter 69: Meridian Mail Trunk Access Restriction

Contents

This section contains information on the following topics:

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 telephone 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 34: Meridian Mail Trunk Access Restriction Call Transfer on page 506, MTAR's capabilities prevent an established call between Telephone A, an internal call, and Telephone B, an external call, from being forwarded to Meridian Mail. When Telephone A attempts to either Transfer, Conference, No Hold Conference or Call Join Telephone B to Telephone 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.

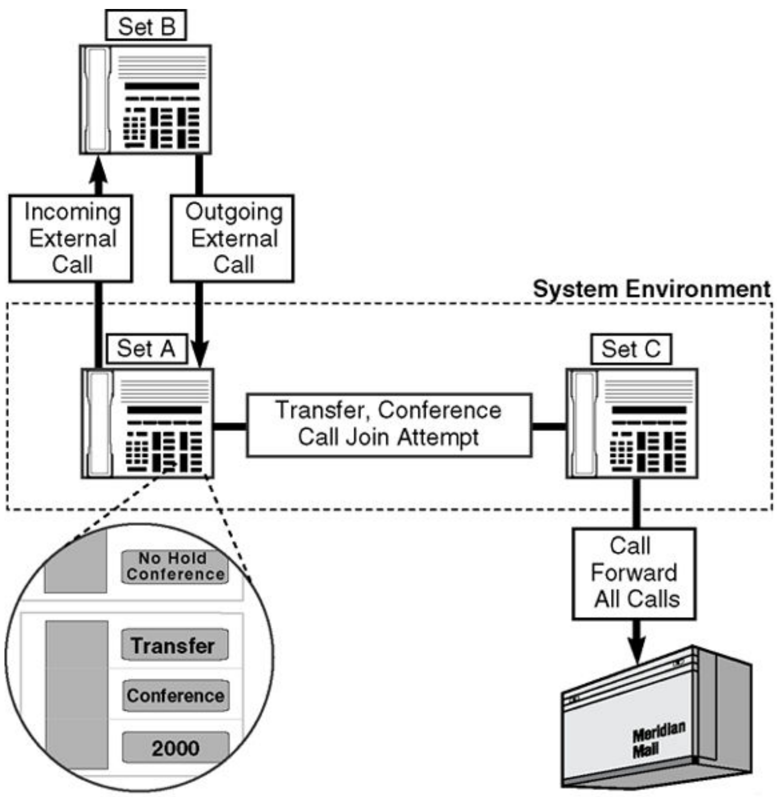


Figure 34: Meridian Mail Trunk Access Restriction Call Transfer

Operating parameters

MTAR does not treat Centralized Attendant Position and Night Attendant telephones as attendant consoles. These telephones receive treatment based on their actual set type. For example, if the night attendant is a proprietary telephone, then it is treated as a proprietary telephone.

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) telephone is not supported.

In a networking environment, Meridian Mail must reside on the same node as the transferring/ conferencing telephone.

Feature interactions

Network Call Transfer, Network Call Conference

Meridian Mail Trunk Access Restriction (MTAR) requires the transferring or conferencing telephone and Meridian Mail to be located on the same node. If the transferring or conferencing telephone 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.

Table 172: LD 15 - Enable Meridian Mail Trunk Access Restriction.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Customer Features and Options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
- OPT	MCI	Message Centre Included.
...		
- MTAR	YES	Meridian Mail Trunk Access Restriction. NO = Meridian Mail Trunk Access allowed.

Feature operation

Call Transfer/Conference

Proprietary telephone

Telephone A is a proprietary telephone with Transfer Key and Conference Key.

1. An incoming/outgoing external call is established between Telephone A and Telephone B, an external party. The call between Telephone A and Telephone B is active on Key X.
2. Telephone A presses the Transfer/Conference Key that automatically puts Telephone B on Hold.
3. Telephone A dials Telephone C. Telephone C has either Call Forward All Calls to Meridian Mail or is a Meridian Mail DN.
4. When Telephone A attempts to transfer/conference Telephone B by pressing more than once the Transfer/Conference Key it is ignored.
5. Telephone A recovers Telephone B by pressing Key X.

No Hold Conference

Proprietary Telephone/ISDN BRI telephone

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.

Telephone A is a proprietary telephone or an ISDN BRI telephone 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 Telephone A and Telephone B, an external party. The call between Telephone A and Telephone B is active on Key X.
2. Telephone A presses the No Hold Conference Key.
3. Telephone A dials Telephone C. Telephone C is has either Call Forward All Calls to Meridian Mail enabled or is a Meridian Mail DN.
4. The conference is Telephone up as normal. However a two party connection between Telephone 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) telephone

Telephone A is an Analog (500/2500-type) telephone with a XFA Class of Service (transfer and three/six party conference allowed).

1. An incoming or outgoing external call is established between Telephone A and Telephone B, an external party.
2. Telephone A performs a switchhook flash that puts Telephone B on hold.
3. Telephone A dials Telephone C. Telephone 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, Telephone A attempts to transfer Telephone B to Meridian Mail by going on-hook.
5. This attempt is treated as an illegal transfer. Telephone A is re-rung and reconnected with Telephone B when going off-hook.

Conference

Analog (500/2500-type) telephone

Telephone A is an Analog (500/2500-type) telephone with a XFA Class of Service (transfer and three/six party conference allowed).

1. An incoming or outgoing external call is established between Telephone A and Telephone B, an external party.
2. Telephone A performs a switch hook flash that puts Telephone B on hold.
3. Telephone A dials Telephone C. Telephone 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, Telephone A attempts to conference Telephone B to Meridian Mail by performing another switchhook flash.
5. The conference is not permitted. Telephone A is reconnected to Telephone B. The call to Meridian Mail is disconnected.

Telephone A is an Analog (500/2500-type) Telephone with a TSA Class of Service (three party service allowed).

1. An incoming or outgoing external call is established between Telephone A and Telephone B, an external party.
2. Telephone A perform a switch hook flash that puts Telephone B on hold.
3. Telephone A dials Telephone C. Telephone 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, Telephone A attempts to conference Telephone B to Meridian Mail by dialing the conference control digits.
5. The conference is not permitted and Telephone A is reconnected to Telephone B. The call to Meridian Mail is disconnected.

Table 173: Summary of Meridian Mail Trunk Access Restrictions on page 511 summarizes how different external calls are handled when Meridian Mail Trunk Access Restriction is enabled.

Table 173: 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

Telephone	External Call Type	Operation	Failure Treatment	Result
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

Chapter 70: Meridian Mail Voice Mailbox Administration

Contents

This section contains information on the following topics:

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 516 and Site with a preconfigured Meridian Mail database on page 524.

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.

Meridian Mail

Although there is no user impact, unsolicited link messages will appear when VMBA is equipped.

Common data elements

Table 174: Data stored by both the system and Meridian Mail on page 515 shows the data that is stored and synchronized between the system and Meridian Mail.

Table 174: Data stored by both the system and Meridian Mail

System	Meridian Mail	Description
DN	Mailbox number	System DN to which a VMB is assigned

System	Meridian Mail	Description
VMB Class of Service	Class of Service	Specific telephone 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 175: Effect of running Audit with Data Correction enabled on page 518.

Table 175: 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 176: Effect of running Upload on page 518.

Table 176: 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

Status of VMB		Effect on VMB databases	
System	**Meridian Mail**	**System**	**Meridian Mail**
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. Table 177: LD 17 on page 520

   Configuring the VMBA application.
2. Table 178: LD 10 on page 521

   Add a VMB on an analog (500/2500-type) telephone.
3. Table 179: LD 11 on page 522

   Add a VMB on a Meridian 1 proprietary telephone.
4. Table 181: LD 20 on page 528.

   Print the DN block

5. Table 182: LD 20 on page 528.

   Print the TN block.
6. Table 183: LD 20 on page 529

   Print VMB data.
7. Table 185: LD 83 on page 530.

   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.

Table 177: LD 17

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.

Prompt	Response	Description
- - 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.

Table 178: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	500 2500	DN related data.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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>.

Prompt	Response	Description
- - 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.

Table 179: LD 11

Prompt	Response	Description
REQ:	NEW	Add.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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. Enabling the VMBA application on page 525
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. Starting a manual audit on page 531

   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. SeeName processing considerations on page 516 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>

- <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>

- <vsid> is the VAS identifier, in the range of 0 to 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>

- <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 180: VMBA application states on page 526

Table 180: 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.

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.

Table 181: LD 20

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.

Table 182: LD 20

Prompt	Response	Description
REQ	PRT	Print.
TYPE	TNB aaaa	TN block, or any telephone configured in LD 11.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.

• Use the PRT VMB command to print the VMB DN and VMB state. For a definition of each state, see Table 184: VMB states on page 529.

Table 183: LD 20

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 184: 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

State	Explanation
	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.

Table 185: LD 83

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 184: VMB states on page 529.

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>

- <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

- <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

- <vsid> is the VAS ID.

Output from this command takes the following format:

```
AUDIT ACTIVE
    x AUDITED y MISMATCHES FOUND/CORRECTED z ERRORS
```

- 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.

   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>

- <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

- <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

- <vsid> is the VAS ID.

Output from this command takes the following format:

```
UPLOAD ACTIVE
    x UPLOADED y DELETED z ERRORS
```

- x is the number of VMBs uploaded
- y is the number of VMBs deleted
- z is the number of failed upload operations

Chapter 71: Message Intercept

Contents

This section contains information on the following topics:

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 telephone 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 telephone 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 telephone is under individual or group Do Not Disturb, indicating the condition.
- Set Status Lockout: when an analog (500/2500-type) or Meridian 1 proprietary telephone goes off-hook, users receive a recorded announcement if any one of the following conditions are met:
    - The Scheduled Access Restriction feature temporarily overrides the telephone Class of Service, Trunk Group Access Restriction, or Network Class of Service restriction level.
    - The Electronic Lockout Feature temporarily overrides the telephone Class of Service or Network Class of Service restriction level.
    - Either the Controlled Class of Service or the Enhanced Controlled Class of Service feature temporarily overrides the telephone Class of Service restriction level.

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 continues until it either times out or the telephone goes on-hook.

Operating parameters

Message Intercept must be specified on a telephone basis.

Comments? infodev@avaya.com

Feature interactions

Announcements

Announcements are not available to attendants or trunks.

The actual announcement received by the telephone is 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. Table 186: LD 56 on page 538

Configure Message Intercept tones.

2. Table 187: LD 10 on page 539

   - Allow Message Interception on an analog (500/2500-type) telephone.

3. Table 188: LD 11 on page 539

   Allow message intercept on a Meridian 1 proprietary telephone.

Table 186: LD 56

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 telephone 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 telephone. 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.

Prompt	Response	Description
- 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.

Table 187: LD 10

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.

Table 188: LD 11

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.

Chapter 72: Message Registration

Contents

This section contains information on the following topics:

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 non-metered 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 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 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 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, see *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

Message Registration (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 telephone, the outstanding PPM is stored against the meter of the telephone. 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. Table 189: LD 16 on page 545

   Activate Message Registration on routes.
2. Table 190: LD 14 on page 546

   Configure the polarity trunk.
3. Table 191: LD 10 on page 546

   Allow or deny analog (500/2500-type) telephones access to meters.
4. Table 192: LD 11 on page 546

   Allow or deny Meridian 1 proprietary telephones access to meters.

Table 189: LD 16

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
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.

Table 190: LD 14

Prompt	Response	Description
REQ	CHG	Change.
TYPE	COT	CO trunks.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	(PIP) PSP	Polarity (insensitive) or sensitive. Use PSP for QPC218, QPC219, QPC295. Use PIP for QPC330, QPC331.

Table 191: LD 10

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	(MRD) MRA	MR (denied) or allowed.

Table 192: LD 11

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	aaa	Digit Display options, where: aaa = ADD, DDS, NDD.
	(MRD) MRA	MR (denied) allowed.

Prompt	Response	Description
KEY	xx MRK	MR key, where: xx = key number.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 73: Message Waiting Indicator by Directory Number

Contents

This section contains information on the following topics:

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 telephones. The MWDN feature provides the following functionalities:

- presentation of multiple message waiting indications on one telephone
- presentation of multiple message waiting indications for one mailbox on more than one telephone
- presentation of remote message waiting indications for message monitoring
- support for one mailbox for multiple Directory Numbers (DNs)

Multiple message waiting indications on one telephone

Prior to the MWDN feature, where more than one DN was configured on one telephone, 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 telephone. The DN associated with the XMWK must be configured as a non-PDN on that telephone.

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 telephone has application for environments where one telephone has the DNs for several individuals.Figure 35: Multiple message waiting indications on one telephone on page 551 shows a scenario where an administrative assistant monitors the DNs for several individuals from the telephone.

Multiple message waiting indications for one mailbox on more than one telephone

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 telephone, the XMWK can be configured for the non-primary appearance of the mailbox DN on each telephone. The DN associated with the common mailbox must be configured as a non-PDN on all the telephones where it appears (except for the one PDN telephone).

When a new voice message is received for the DN associated with the common mailbox, all the XMWKs configured on all the telephones 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.

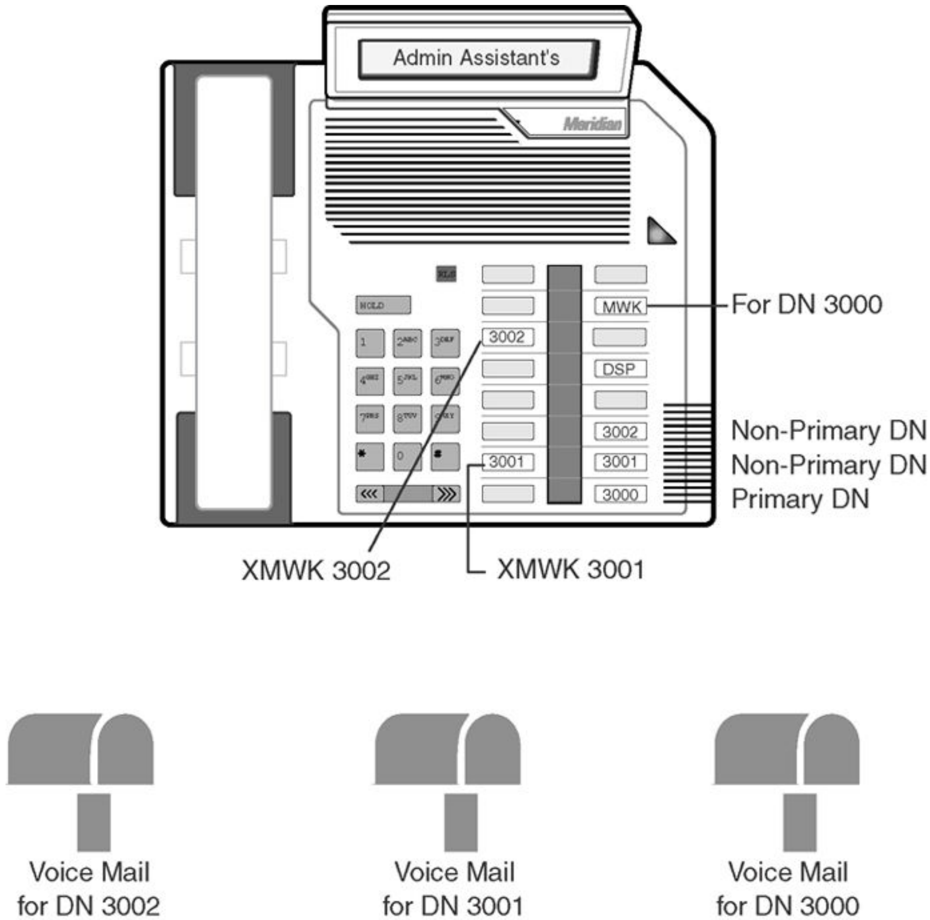


Figure 35: Multiple message waiting indications on one telephone

Figure 36: Multiple message waiting indications for one mailbox on more than one telephone on page 552 shows multiple message waiting indications for one mailbox on many telephones. An application of this component of the MWDN feature is a person with more than one telephone 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 telephone and their mobility telephone. In this scenario, both the desk telephone and mobility telephone must be on the same switch.

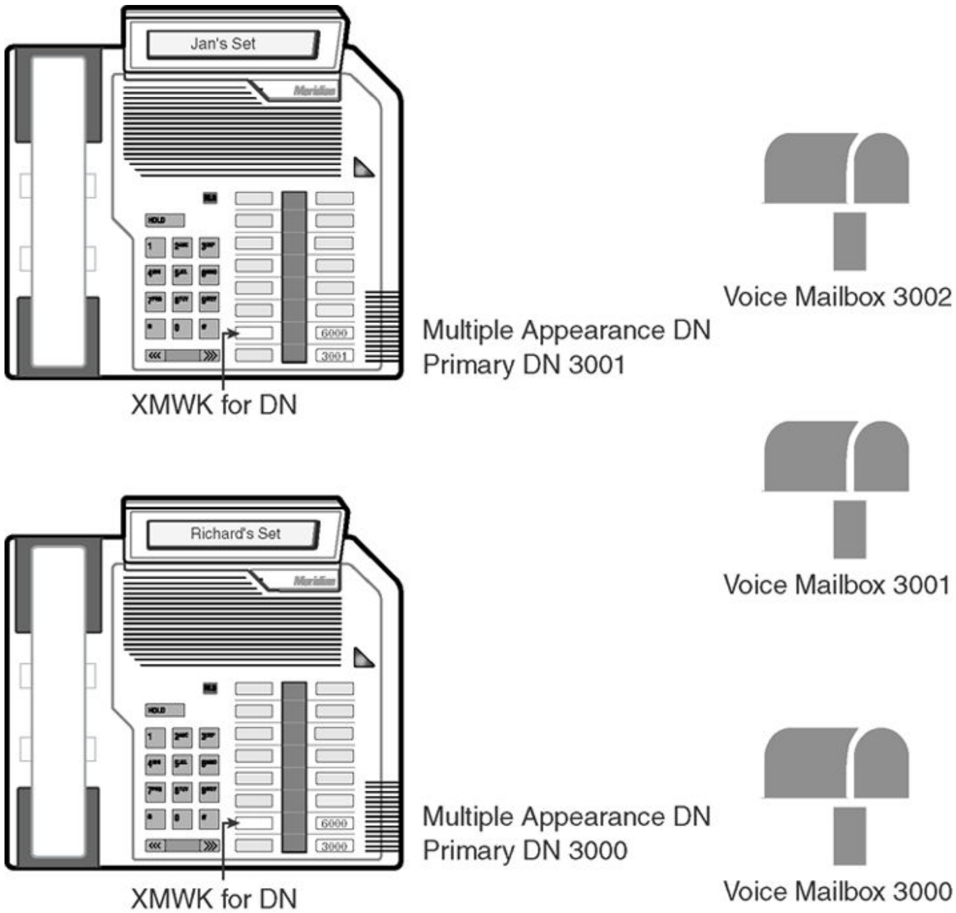


Figure 36: Multiple message waiting indications for one mailbox on more than one telephone

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 telephone to query if they had voicemail. With the MWDN feature, users can monitor the status of their mailboxes from a remote telephone without logging into their telephones. 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 telephone 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 telephone.

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 37: RMWK operation when a Phantom TN is call forwarded to a telephone on page 554 illustrates a Phantom TN, DN 3001, with a RMMA/RMMO class of service. A RWMK key is configured on a telephone 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 telephone 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 38: One mailbox for multiple DNs on page 555 shows DNs 5000, 5001 and 5002 with a shared mailbox.

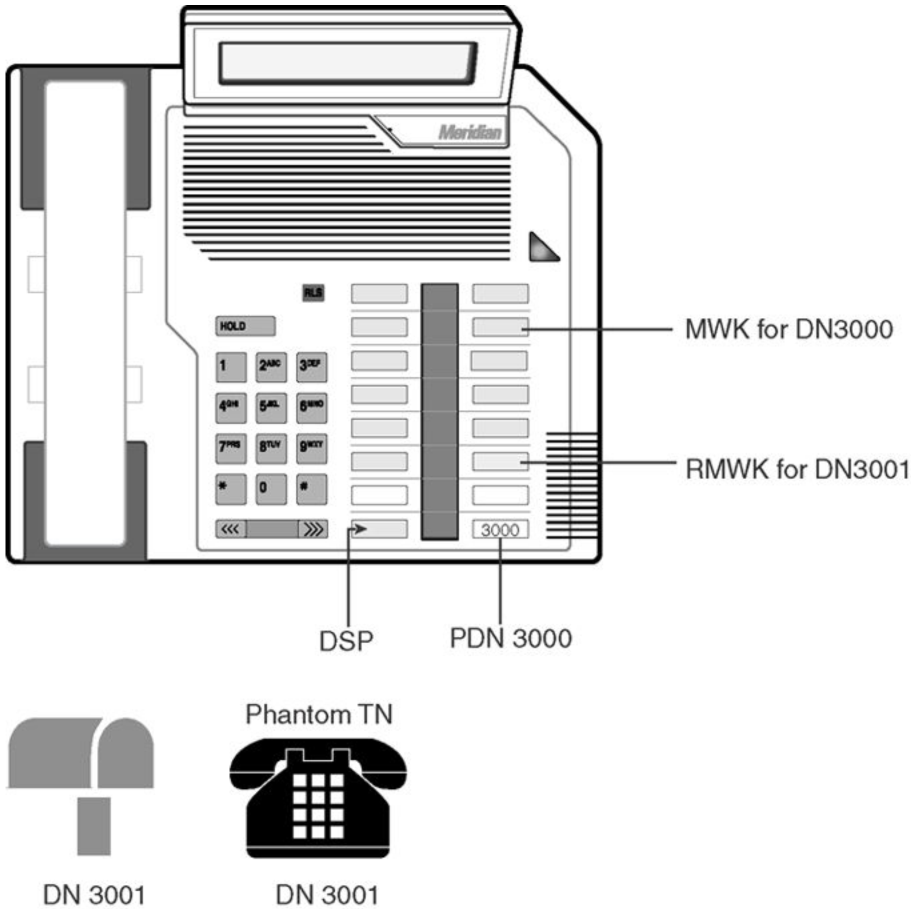


Figure 37: RMWK operation when a Phantom TN is call forwarded to a telephone

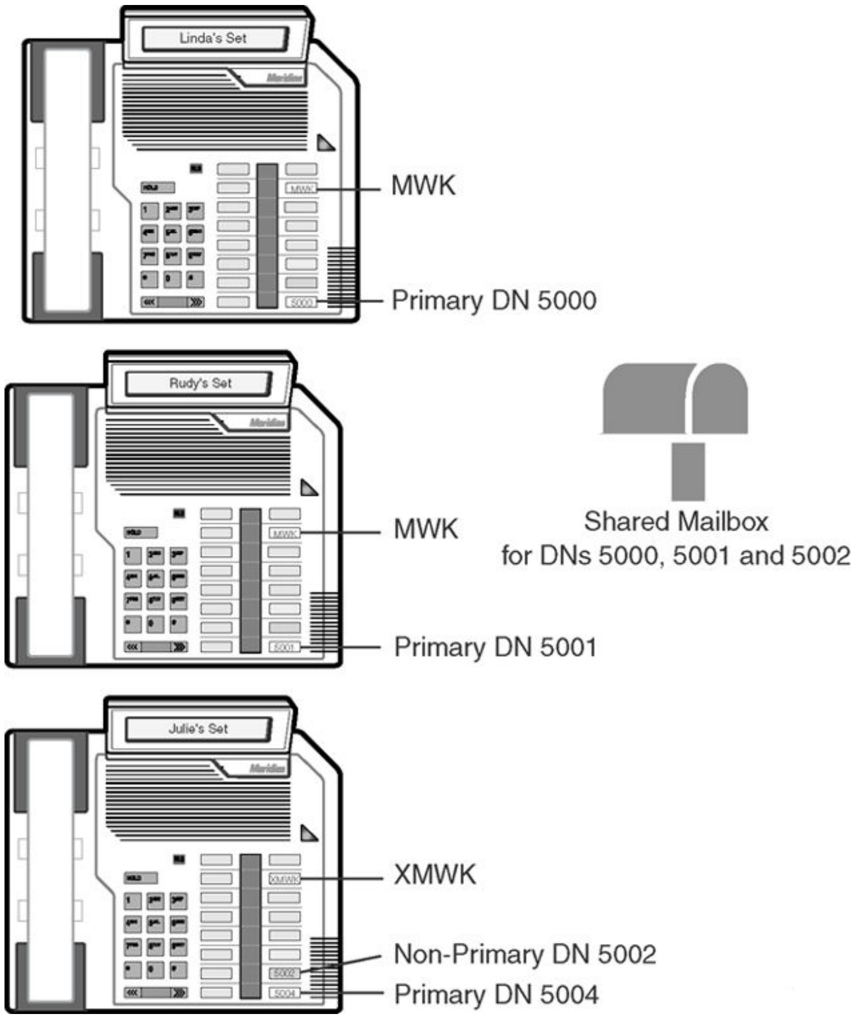


Figure 38: One mailbox for multiple DNs

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 telephone.

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 telephones where the DN is configured as a PDN, the message waiting indication occurs on the MWK and red LED.
- On telephones 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 telephone. With the MWDN feature, a user can monitor the mailbox associated with the Phantom DN through the RWMK on a proprietary telephone.

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. Table 193: LD 15 on page 558

   Enable the Message Center in the Customer Data Block
2. Table 194: LD 11 on page 559

   Configure the class of service option for the XMWK
3. Table 195: LD 11 on page 559

   Configure the Remote Message Waiting Key (RMWK) to monitor remote telephones with a mailbox.
4. Table 196: LD 10 on page 560

   Configure new class of service options for analog (500/2500-type) telephones.
5. Table 197: LD 11 on page 561

   Configure the class of service options for Meridian 1 proprietary telephones.
6. Table 198: LD 10 on page 561

   Extend the message waiting indication function to all the analog (500/2500-type) telephones.
7. Table 199: LD 11 on page 562

   Extend the message waiting indication function to all the Meridian 1 proprietary telephones

For all the following tasks, first enable the Message Center in the Customer Data Block in LD 15. See Table 193: LD 15 on page 558.

To configure multiple message waiting indications on one telephone or on many telephones:

- Configure the class of service options for the Extended Message Waiting Key (XMWK) and its LCDs on proprietary telephones in LD 11. See Table 194: LD 11 on page 559.

To configure remote message waiting indications on one telephone:

- Configure the Remote Message Waiting Key (RMWK) to monitor remote telephones with a mailbox in LD 11. See Table 195: LD 11 on page 559.
- Configure new class of service options to enable analog (500/2500-type) telephones to be monitored remotely in LD 10. See Table 196: LD 10 on page 560.

  All telephones with primary DNs to be monitored must be configured as RMMA/RMMO.
- Configure new class of service options to enable proprietary telephones to be monitored remotely in LD 11. See Table 196: LD 10 on page 560.

  All telephones with primary DNs to be monitored must be configured as RMMA/RMMO.

To configure message waiting indications on telephones where DNs sharing a mailbox appear:

- Configure the class of service option to enable the message waiting indication for all the analog (500/2500-type) telephones on which the DNs sharing the mailbox are configured in LD 10. See Table 198: LD 10 on page 561.
- Extend the message waiting indication function to all the Meridian 1 proprietary telephones on which the DNs sharing the mailbox are configured in LD 11. See Table 199: LD 11 on page 562.

Table 193: LD 15

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 telephone.

The DN associated with the XMWK must not have an XMWK already associated with it on this telephone. Use Table 194: LD 11 on page 559 to configure the class of service option for the Extended Message Waiting Key (XMWK) and its LEDs on the proprietary telephones.

Table 194: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied.
	LMPX	The red LED on the proprietary telephones 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 telephones 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 XWMK cannot be configured on key 0.

Table 195: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. (MWD) = Message Waiting Denied.
	LMPX	The red LED on the proprietary telephones 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).

Prompt	Response	Description
		(LMPN) = The red LED on proprietary telephones 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}

All telephones with primary DNs to be monitored must be configured as RMMA/RMMO.

Use Table 196: LD 10 on page 560 to configure a new class of service options to enable analog (500/2500-type) telephones to be monitored remotely.

Table 196: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	500	Analog (500/2500) telephone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
...		
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	RMMA RMMO	Allow the telephone to be remotely monitored for messages. Allow the telephone to be remotely monitored for messages and allow the telephone to override, if it is being monitored already. (RMMD) = Deny telephone for Remote Monitoring of Messages.

All telephones with primary DNs to be monitored must be configured as RMMA/RMMO.

Use Table 197: LD 11 on page 561 to configure the class of service to enable Meridian 1 proprietary telephones to be monitored remotely.

Table 197: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS		Class of service option.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	LMPX	Enable the red LED on the supported proprietary telephones 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 telephones to reflect the status of the mailboxes associated with both PDN and non-PDNs.
	RMMA RMMO	Allow telephone for Remote Monitoring of Messages. Allow telephone for Remote Monitoring of Messages and Override, if it is being monitored already. (RMMD) = Deny telephone for Remote Monitoring of Messages.

Voice Mailbox Administration (VMBA) must be configured before configuring one mailbox supporting Multiple Appearance DNs. For information about configuring VMBA, see *Hospitality Features Fundamentals, NN43001-553*.

Use Table 198: LD 10 on page 561 to extend the message waiting indication function to all the analog (500/2500-type) telephones on which the DNs sharing the mailbox are configured.

Table 198: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	500	Analog (500/2500) telephones.
...		
CLS		Class of service option.

Prompt	Response	Description
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.
	SMWA	Allow Extended Message Waiting Indication. (SMWD) = Deny Extended Message Waiting Indication.

Voice Mailbox Administration (VMBA) must be configured before configuring one mailbox supporting Multiple Appearance DNs. For information on configuring VMBA. see *Hospitality Features Fundamentals, NN43001-553*.

Use Table 199: LD 11 on page 562 to extend the message waiting indication function to all the Meridian 1 proprietary telephones on which the DNs sharing the mailbox are configured (whether the DNs are PDN or non-PDN).

Table 199: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
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 telephone in idle position.
2. The telephone 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 telephone displays RMWK.
3. Enter the new mailbox DN.
4. The screen displays the digits. Press the RMWK to validate the mailbox DN.

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 telephone on which this DN is configured as PDN has a class of service telephone 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 telephone on which this DN is configured as PDN has a class of service set to RMMO and is being monitored by another telephone, this telephone overrides and continues to monitor.
    - Overflow tone is given if any telephone on which this DN is configured as PDN has class of service set to RMMD.
    - Overflow tone is given if any telephone on which this DN is configured as a PDN has class of service set to RMMA and is being monitored by another telephone.
6. To cancel remote message monitoring, press the RMWK when it is lit or flashing.

Chapter 74: Message Waiting Lamp Maintenance

Content

The following are the topics in this section:

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

Table 200: 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.

Chapter 75: Message Waiting Unconditional

Contents

This section contains information on the following topics:

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 analog (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 analog (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 because 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 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

Table 201: 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).

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.

Chapter 76: Mixed disk drive sizes with CP PII

Contents

This section contains information on the following topics:

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

Because no pack ID is available, there is no data provided to the Inventory Reporting Feature.

CS 1000M SGand Meridian 1 PBX 61C

The CS 1000M SG and Meridian 1 PBX 61Csystems 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.

Chapter 77: Mobile Extensions

Contents

This section contains information on the following topics:

Conventions

This section uses the following conventions:

- Mobile Extensions and Mobile X will be used interchangeably
- a Communication Server 1000M (CS 1000M) and a Communication Server 1000E (CS 1000E) are referred to generically as a CS 1000 system

Feature description

Hardware and software requirements

The hardware and software requirements for the Mobile Extensions feature are as follows

- CS 1000 CP PIV or CP PM system
- CS 1000 Release 5.5 or later
- a mobile telephone
- Mobile X (MOBX) package 412 – for configuring Mobile X for mobile users
- Personal Call Assistant (PCA) package 398 – for programming dialing information to reach the user on the mobile telephone and for basic operations of the Universal Extension
- ISDN package 145 and associated ISDN/Networking packages – for configuring the Mobile X network routes
- UEXT
- ISDN PRI trunk

Functionality

The CS 1000 Mobile Extensions feature enables mobile telephones connected to a CS 1000 system to appear as office extensions. Incoming calls to an office telephone number automatically call out to the mobile telephone associated with the office telephone number. A user making a call from a mobile telephone can use the enterprise dial plan as if the call were made from an office extension. Call treatments provided for an office extension are extended to the mobile telephone.

A Mobile user is configured with a Universal Extension (UEXT), which provides the logical connection to the mobile telephone. A Mobile user can have an office desktop telephone (for example, digital, IP) in addition to the Universal Extension. A single Directory Number (DN) can be shared between the Universal Extension and the desktop telephone.

The CS 1000 Mobile Extensions feature provides the following business grade telephony features to mobile telephone users:

1. A call to the shared DN will ring the mobile telephone and desktop telephone simultaneously. The call can be answered from the mobile telephone or from the desktop telephone.
2. A call from the mobile telephone can use the enterprise dialing plan as if the call were made from an office extension. The call will be presented to the called party as if the call were made from an office extension. The Calling Line ID and the Calling Party Name will be that of the office extension.
3. If the mobile telephone is in use, a busy status is indicated on the desktop telephone.
4. Call Forwarding features such as Call Forward All Calls, Internal Call Forward, Call Forward Busy and Hunting are applicable to the user's directory number. Call Forward features can be activated from the mobile telephone.
5. Call Management features such as Group Hunt, Intercept Computer and Call Party Privacy can be activated/deactivated from the mobile telephone.
6. Call restrictions are applied to calls made from the mobile telephone in the same way as calls made from an office extension, using the Class of Service (CLS) of the Universal Extension.
7. Attendant features such as Camp On and Busy Lamp Field indication are applicable to mobile telephones.
8. The Attendant can Barge-in to a call that the user has answered on the mobile telephone and the associated warning tone is provided to the mobile telephone.
9. TN type Universal Extension (UEXT), is used to facilitate CS 1000 mobility capabilities. This allows Mobile X Line, Fixed Mobile Convergence Line, SIP Line and Telephony Services User to be configurable on a per-user basis.
10. If the Mobile X user has a regular office desktop telephone with a prime key sharing the same DN in a Single Call arrangement, the Mobile X user can handoff a call between the desktop telephone and the mobile telephone. A security password must be entered, if enabled, before being able to take a call from the mobile telephone.
11. While the Mobile user on the mobile telephone is talking with another party, the user can initiate a second call to transfer, conference, toggle, cancel or complete the second call using the Mid-Call features. This feature is invoked by using the Mobile Feature Activation Code (MFAC) followed by the Flexible Feature Code (FFC) in a traditional User Interface (UI) and by dialing MFAC and MPO control codes in a simplified UI.
12. When a Mobile X user makes a call from a mobile telephone to an Enterprise Network DN, the call is routed to the CS 1000 system with a dedicated mobile route. This is known as source based routing, as the Service Provider routes the call to a particular switch based on the source of the call. The trunk interface of this mobile route is supported on most of the ISDN interfaces including EURO, ESIG, QSIG, MCDN and NI2.

    With source based routing, mobile initiated calls are routed to the mobile user's home PBX regardless of what the user dials on the mobile telephone. Exceptions are emergency numbers (for example, E911) and other Mobile services. In other words, the mobile network will route the mobile initiated call based on a pre-defined arrangement for the mobile telephone and not based on the dialed digits.

A dedicated route is only permitted to carry Mobile X traffic inbound. All calls presented to the PBX on the route with MBXR are required to provide a CLID that matches the UXID of a Mobile X UEXT phone. All other calls are blocked.

The dedicated route can be used for any outbound traffic. This provides the customer the opportunity to off load some of their outbound traffic from regular PRI loops that also carry inbound calls. They can choose to make the Mobile X route the first choice route for some outbound calls, which reduces traffic on another bidirectional span. Do not overload the route with outbound calls or the Mobile X users can be blocked from making their incoming calls.

13. As an alternative interface to source based routing, a Mobility Service Access interface using a Mobile Service DN is used to allow customers to setup a connection between the PBX and a standard Central Office. The CO provides the connectivity to the Mobile Network. This interface enables the mobile user to use FFCs to access Mobile X features. Mobile Service Access is only supported for PRI trunks in CS 1000 Release 5.5.
14. A call on a dedicated Mobile Route, or a call through Mobile Service Access, requires matching and validation of the CLID from the mobile telephone. The CLID received from the mobile telephone must match the UXID configured for a Universal Extension in order for the call to go through. Attributes of the call are taken from the matched UEXT.
15. An ISM is used for the MOBX type of UEXT. MOBX Mobile X telephones will also consume a TN ISM. The CS 1000 Mobile Extensions capability is managed with a Mobile X package and a Mobile X ISM.
16. Trunk call recorder is used to record the Mobile X calls. Trunk call recorder records a call by monitoring the trunks that connects the Communication Server 1000 system to the Mobile Network.

Mobile Extensions system overview

The following diagram depicts the major components of a Mobile Extensions system.

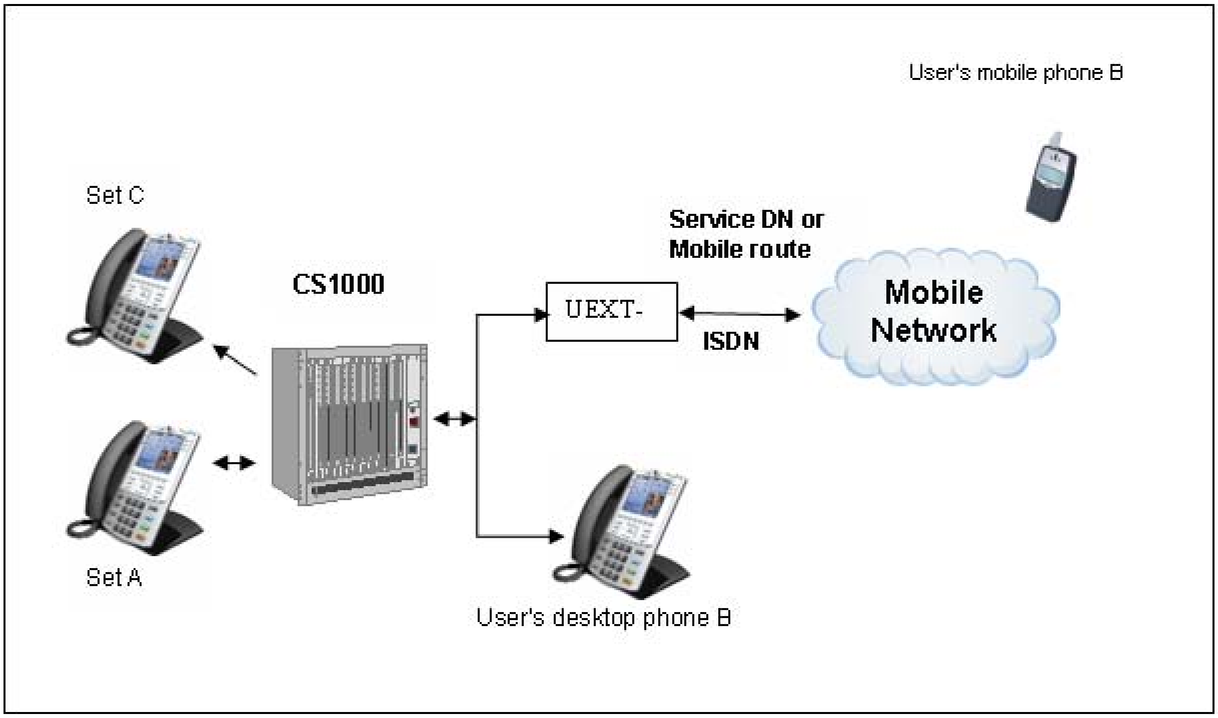


Figure 39: Mobile Extensions system overview

The major components of a Mobile Extensions system consist of:

1. A dedicated route that is used to connect the PBX and a standard Central Office. The CO provides the connectivity to the Mobile Network. The Mobile Network routes all calls (except emergency or special service calls) made from the mobile telephone to the CS 1000 system through the dedicated route. The dedicated route can also be used to make calls to the mobile telephones.
2. A Mobile Service Access DN that can be dialed from the mobile to connect the Mobile Network to the CS 1000 system. The Mobile Service Access DN can be used to make a call to another party within the Enterprise Network, or to activate Call Management features using FFCs.
3. A Universal Extension Mobile X telephone configured for each Mobile user with a Hot P key programmed with dialing information to reach the user's mobile telephone. The Mobile UXID (which must be the same as the CLID for an incoming call from the mobile telephone) is also programmed on the Universal Extension.
4. An optional desktop telephone for each user, which shares the same DN as the Mobile X Universal Extension.
5. An internal table that is built when Mobile X Universal Extensions are configured during system initialization. The internal table is used to determine which Mobile user originates a call by matching the mobile telephone DN to the Mobile user's internal DN.

When a call is made from the mobile telephone, the Mobile Network will route the call to the CS1000 system that has the Mobile user's Universal Extension defined. The CLID of this call will be the mobile telephone DN. By searching through the internal table, a match of the mobile telephone DN to the corresponding internal DN can be made.

Call Arrangements

A Single Call Arrangement is possible to associate the Mobile user's Mobile X and the user's desktop telephone. This arrangement provides the mechanism to identify that a user's DN is busy, regardless of whether the user is busy on the mobile telephone or is busy on the desktop telephone. The operation of Mobile X telephones differs from PCA telephones, in that the Mobile X keeps track of the busy status of the associated external device, with respect to calls made to it through the Mobile X. This allows the CS 1000 Call Server to provide Busy Treatment when the device is busy. In a Single Call Arrangement only one call can be made from the DN at a time. This means that if the desktop telephone is involved in a call using the shared DN, the Mobile User will not be able to make a call. The CS 1000 Call Server will give an overflow tone when the call is connected through the Mobile Route or Mobile Service Access.

A Multiple Call Arrangement can also be used for Mobile Extensions. With this configuration the two telephones will be able to make or receive calls, independent of the busy status of the other device. The busy status of the UEXT is still tracked for calls involving the mobile telephone, but a call on the mobile telephone will not prevent a call being made to the shared DN from terminating on the desktop telephone, and vice versa. This arrangement may be preferable in cases where the desktop telephone could be used by someone other than the usual user of the shared DN, since otherwise the Mobile User may be blocked from making a call when the desktop telephone is in use.

Call to the Enterprise DN

Basic Operation

When a call is made to the Mobile user's enterprise DN the CS 1000 system presents the call to the user's desktop telephone (if equipped) as well as the user's Mobile X. The Mobile X makes use of the HOT P key to call out to the user's mobile telephone.

The HOT P key for the UEXT has the dialing information for calling the user's mobile telephone (i.e., "Access Code to reach the Mobile Network" plus the "mobile telephone number"). The Access Code must be programmed to make the call to the Mobile Network via an ISDN PRI trunk.

The following diagram depicts the basic operation when both telephones are idle.

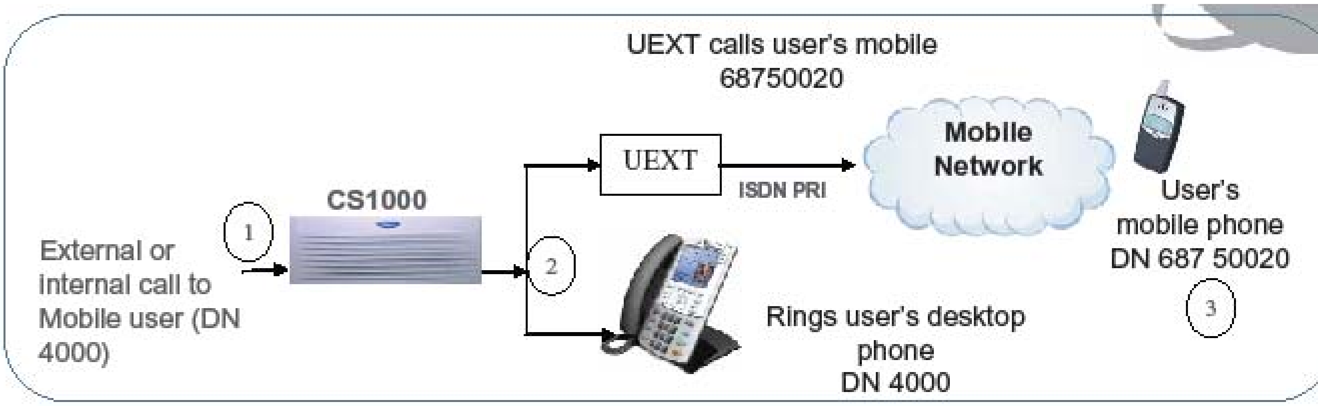


Figure 40: Call to the enterprise DN

Both telephones are idle

When a call is made to the Mobile user's enterprise DN and both the desktop telephone and the mobile telephone are idle, both the desktop telephone and the mobile telephone are rung.

The call can be answered from the desktop telephone or from the mobile telephone. If the telephones are in a single call arrangement the user's enterprise DN is busy, otherwise the specific telephone that answered is marked busy for this DN.

If the call is answered from the desktop telephone, the mobile telephone stops ringing. In a single call arrangement there is no busy status indication shown on the mobile telephone, but calls cannot be made to or from the mobile telephone.

One telephone in a multiple call arrangement is busy

When some telephones in a multiple call arrangement are busy, but some are idle, the call is presented to the idle appearances. Busy appearances are unchanged. The call can be answered from any telephone that is ringing. Once the call is answered the other ringing extensions become idle.

A DN is busy

When any appearance in a single call arrangement is busy on the DN or all appearances in a multiple call arrangement are busy the new call is given a busy treatment. The call treatment, depending on the configuration of the MARP TN (Multiple Appearance Redirection Prime TN), could be hunting to another DN, forward to attendants, forward to Voice Mail, or simply giving back a busy tone.

Call made from mobile telephone

Basic operation

When a call is made from the mobile telephone according to the enterprise dial plan format, the Mobile Network routes the call to the CS 1000 system where the user information is configured. The CS 1000 system analyzes and connects the call. The following diagram illustrates the call flow of a sample call and the sequence of events that take place.

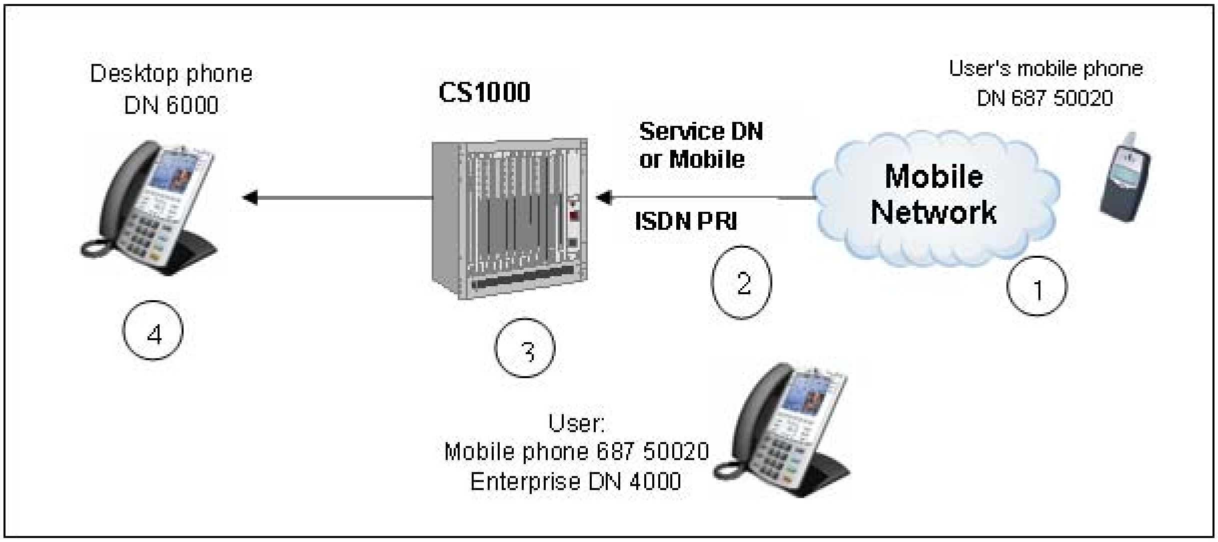


Figure 41: Call from the mobile telephone

1. The user makes a call from the mobile telephone and dials the destination DN (i.e. 6000) as if the user is making the call from the office.
2. The Mobile Network routes the call to the CS 1000 system where the Mobile user is configured. The call is made either through the dedicated ISDN PRI route or a Mobile Service Access DN. The CLID of the call will be the DN of the mobile telephone (for example, 68750020). The called party DN is 6000. When the call is made through the Mobile Service Access DN the user will dial the MSA DN, will hear dial tone, and then dial the desired DN.
3. The CS 1000 system tries to match the CLID to the user’s enterprise DN by examining its internal database. Once a match is found, the call will be treated as if it is made internally from the user’s enterprise DN. In this example, the mobile telephone DN 68750020 is found to be associated with user’s DN 4000.
4. The destination set is then rung. The display on the destination set shows the user’s enterprise DN (4000) and the corresponding name for DN 4000. The user's enterprise DN appearance is marked busy. In a single call arrangement this will make the DN busy.

Note:

If a match for the CLID is not found, the incoming trunk call is blocked.

Source based routing

There are two possible configurations of the Mobile Extensions trunk routing, with or without a dedicated mobile route to the PBX. With a dedicated route, when a Mobile X user makes a call from a mobile telephone the call is routed to the CS 1000 system on the dedicated route, based on the originator's identification. This is known as source based routing, as the Service Provider routes the call to a particular switch based on the source of the call. All incoming calls on the dedicated route are considered as mobile and the Mobile Extension specific features (such as CLID and NCOS enhancements and Busy Status) are applied automatically.

With Source based routing, mobile initiated calls are routed to the mobile user's home PBX regardless of what the user dials on the mobile telephone. Exceptions are emergency numbers (for example, E911) and other Mobile services. In other words, the mobile network will route the mobile initiated call based on a pre-defined arrangement for the mobile telephone and not based on the dialed digits.

Source based routing is the preferred interface for the mobile operator to route a call from a mobile telephone to CS 1000 system. If source based routing is not available the mobile user must dial the Mobile Service Access DN first to gain access to the CS 1000 system. ISDN PRI routes are used to connect to the Mobile Network, for both source based routing and Mobile Service Access. The trunk interface of the mobile route is supported on most of the ISDN interfaces including EURO, ESIQ, QSIG, MCDN and NI2.

Mobile Service Access DN

When it is impossible to separate the mobile calls from other external calls and mark the single route as mobile, the Mobile Service DN can be configured for these purposes. The Mobile Service DN allows the CS 1000 system to detect the incoming call as a call from the mobile telephone. Once the call reaches the Service DN, the CS 1000 system looks up the Caller ID against the database and allows or rejects the incoming call. After that dial tone is provided and the Service DN allows the Mobile X user to gain access to the CS 1000 system. The Mobile user may dial an enterprise DN, or activate the Mobile X Call Management features using FFCs.

Mobile Service Access DNs will not operate as described if called from an IP trunk, if the digits from the trunk are not received as in-band DTMF tones.

When a Service DN is used to activate a Call Management feature from a mobile telephone, a FFC is used after the call is connected to the CS 1000 Call Server. The FFC features supported on the mobile telephone for Call Management are listed in the following table:

Table 202: FFC used for Call Management features for Service DN call

FFC Abbreviation	FFC meaning
CFWA, CFWD, CFWV	Call Forward Activate, Deactivate, and Verify
ICFA, ICFD, ICFV	Internal Call Forward Activate, Deactivate, and Verify
PCAA, PCAD	PCA Activate and Deactivate
ICPA, ICPD	Intercept Computer Activate and Deactivate
ICPP	Calling Party Privacy
IGHTA, GHTD	Group Hunt Activate and Deactivate
ICFHO	Call Forward Hunt Override
|RPAX, RPAN	Dial and answer RPA (radio paging) calls

Mobile Extensions DTMF Tone Monitor

While a mobile telephone is in use with an enterprise DN within a CS 1000 system, there is a need for the Mobile X user to signal the PBX for activating Call in Progress features such as Call Transfer and Conference. The signal initiated by the mobile telephone is transmitted in-band as part of the voice path. The Mobile X DTMF Tone Monitor is a CS 1000 Call Server software feature for extracting the Dual Tone Multifrequency (DTMF) tone from the voice path of the mobile telephone. The Mobile X DTMF Tone Monitor makes use of the VGW resources in the CS 1000 system to listen to all mobile trunks with active calls. When a specific Mobile Feature Activation Code (MFAC) is detected by the DTMF tone monitor, the VGW attached to the mobile trunk signals the CS 1000 system on behalf of the mobile telephone for further actions. The MFAC is not valid in normal dialing sequences and therefore does not have to be valid within the customer’s dialing plan.

Once the MFAC is detected by the VGW resource the CS 1000 Call Server will disconnect the VGW port, provide Dial Tone and detect the remaining DTMF digits for Call in Progress feature activations through standard DTR resources, similar to Direct Inward System Access.

When the voice path is monitored for DTMF the user may begin to perform End to End signaling, perhaps to provide information to an application such as an IVR or Voice Mail. Since the digits required for the application may conflict with the MFAC, the software will stop monitoring the voice path for the MFAC if it detects digits that do not match the MFAC. The monitoring for MFAC will remain disabled for the duration of the call, unless the call is placed on hold and then retrieved by the other local CS 1000 party in the call.

Mobile X Feature Activation Code (MFAC)

When a Mobile X user's mobile telephone is connected to another telephone in the CS 1000 system, the user can initiate Call in Progress features such as Call Transfer or Conference.

The Mobile X user must press a special key sequence Mobile X Feature Activation Code (MFAC) to signal the CS 1000 system for the feature activation. This special key is configured as the MFAC in Overlay 15 (see Feature implementation on page 601).

The MFAC is not valid in normal dialling sequences and therefore does not have to be valid within the customer's dialling plan.

Some Mobile providers may use the "*" (the recommended MFAC), as the initiation of a Mobile Network feature. Coordinate with the Mobile Network provider and the Public Network provider to ensure this PBX configuration is not conflicting with their feature interface.

Mobile X DTMF Tone Monitor Operation

In order to provide features on an active call on a mobile telephone the system connects a DTMF detector to listen to the voice path from the mobile telephone. When the mobile user wishes to activate transfer or conference he presses the digits configured as the MFAC. (This MFAC does not have to be unique within the customer's dial plan, since it is only dialed on an active call.) Once the MFAC is detected, the monitor stops the scanning of this trunk and special dial tone is provided to the mobile telephone. A DTR is used to listen to the Mobile user for more digits of FFCs to determine further actions.

The major components of a Mobile X DTMF Tone Monitor consist of:

1. Attaching/Detaching the DTMF device
2. Extracting the DTMF tone for MFAC
3. Call in Progress Feature Notification
4. Detecting conditions to stop monitoring for DTMF

Call in Progress Features

After an active call is established with a Mobile X user, the Call Transfer, Conference, Toggle, cancel, or Complete Call in Progress features can be activated from the mobile telephone by using a DTMF signal. These features are also called Mid-Call features. The Mobile X user can dial the MFAC to signal the CS 1000 system to initiate a Call in Progress feature.

The Call in Progress features (sometimes called Mid-Call features) supported for Mobile X are as follows:

- Conference
- Transfer (Blind and Consultation)
- Toggle
- Cancel
- Complete

Following are the two User Interfaces (UIs) to activate the Mid-Call features:

- Traditional UI
- Simplified UI.

Only one interface can be configured for one customer (Overlay 15).

Device Handoff

This feature allows the Mobile X user to hand off an established call between the mobile telephone and the desktop telephone. The desktop telephone must be configured with a Handoff key.

This feature can only be used when the desktop telephone and the Universal Extension share the same Primary DN.

A Conference circuit is required to perform this operation. If no conference device is available the key will not function.

The Handoff feature only supports a single mobile phone UEXT unit. It is not possible to perform the Handoff feature for several mobile sets.

Handoff from Mobile telephone to Desktop telephone

When a call is established on the mobile telephone, the Mobile X user can hand off the call to the desktop telephone in a single call arrangement. To prevent unauthorized handoffs the desktop telephone user must dial the Station Control Password (SCPW) of the UEXT telephone, unless the POA class of service (Privacy Override Allowed) is configured for the desktop telephone.

Device Handoff only applies to Single Call DN arrangements. Multiple Call DNs may transfer a call between appearances.

Handoff from Desktop telephone to Mobile telephone

When the call is established on the desktop telephone, the Mobile X user can handoff the call to the mobile telephone.

Feature interactions

Blind Transfer

To activate the Blind transfer feature, a mobile telephone user dials the Transfer-init FFC with the DN where the user wants to transfer the call. Upon hearing a ring tone, the user can complete the transfer by dialing MFAC and the Feature-complete FFC (blind transfer). If the DN is in the busy state, or unequipped, a busy or overflow tone will be provided to the user for 2 seconds and the user will be reverted back to the initial call.

Call Management features

Attendant Barge-in

The Attendant Barge-In (BIN) feature allows the Attendant to establish a connection with any trunk in the system to verify that the trunk is in working order. When Attendant 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.

Boss/Secretary Filtering Enhancement

A subset of the Boss/Secretary Filtering Enhancement features can be applied to the Mobile X arrangement. The mobile telephone can be used as a boss telephone, but not as a secretary telephone. The boss telephone's BFS key can not be activated from the mobile telephone. Therefore, only the BSFE features that are controlled from the secretary telephone are available.

The BFS key for the secretary telephone has to be configured with the mobile user's UEXT TN. With this configuration, the secretary can:

- Activate and deactivate call filtering for the mobile user (boss)
- Transfer an incoming call from the secretary to the mobile telephone (boss)

Busy lamp field indication on Attendant console

The Busy Lamp Field feature can be used to monitor the state of the Mobile user's mobile telephone. If the Mobile user's enterprise DN is configured for BLF on the Attendant console, the lamp state on the BLF module will reflect the state of the user's UEXT (which corresponds to the state of the mobile telephone).

Call Forward All Calls

If Call Forward All Calls is activated on the user's enterprise DN, any calls made to the user's DN will be call forwarded to the CFW DN regardless of the state of the user's desktop telephone or the user's mobile telephone.

Call Forward All Calls can be activated/de-activated on the user's desktop telephone. It can also be activated/de-activated from the mobile telephone using a FFC. The Mobile user first must dial the Mobile Service DN to establish an ISDN trunk connection between the CS 1000 system and the Mobile Network. After an ISDN trunk connection is established, the mobile telephone user dials the CFW FFC.

Call Forward and Busy Status (BFS)

The Call Forward and Busy Status (BFS) feature can be used on the mobile telephone. The BFS key for the monitoring party has to be configured with the Mobile user's UEXT TN, rather than with the Mobile user's desktop TN.

With this configuration, the monitoring party can:

- Monitor, activate and deactivate Call Forward for the Mobile user
- Override Call Forward of the Mobile user, in order to place a call to the Mobile user, and
- Determine whether the Mobile user is busy on a call using the mobile telephone

Call Forward Busy

If Call Forward Busy is configured on the user's UEXT, when the mobile telephone is busy DID calls to the user's enterprise DN will be forwarded to the Attendant. For Multiple Appearance DN, the MARP TN determines the operation of call redirection.

Note:

On incoming DID calls, Hunting takes precedence, followed by Call Waiting, then Call Forward Busy.

Call Forward Hunt Override

A mobile user who makes a call and detects that it has been redirected can make a call through the MSA DN and dial the CFHO FFC and then the target DN, in order to insure that the call is terminated on the desired DN and not redirected.

Call Forward, Internal Calls

The Internal Call Forward feature allows the user to selectively forward only internal calls to the Internal Call Forward DN. If Internal Call Forward is activated for a Mobile user, all internal calls are forwarded to the ICF DN.

Call Forward Internal Calls can be activated/de-activated on the Mobile X DN from the mobile telephone using the ICFA/ICFD FFC. The Mobile user first must dial the Mobile Service DN to establish an ISDN trunk connection between the CS 1000 system and the Mobile Network. After an ISDN trunk connection is established, the mobile telephone user dials the ICFA/ICFD FFC. The ICF DN can also be changed using the ICFA FFC.

For Multiple Appearance DN, the MARP TN determines the operation of call redirection.

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.

Call Forward No Answer

If a call made to the user's mobile telephone is not answered after a configured number of ring cycles, the call forwards based on the configuration of the MARP TN. The MARP TN can be the desktop TN or the UEXT TN. CFNA can be configured to forward to a voice mail system, Attendant, or another extension.

Important:

To compensate for the short delay before the call rings the mobile extension, it is recommended that you configure the CFNA ring count to a minimum of 4 rings in LD 15.

Call Hunting

If Call Hunting is allowed on the UEXT and the HUNT DN is defined, any calls to the user's enterprise DN when the user is busy, are hunted to the DN defined in the UEXT data. If the user also has a desktop telephone, the MARP TN determines the operation of call redirection.

Calling Party Privacy

The Calling Party Privacy (CPP) feature enables the CS 1000 system to block the display of the Calling Party's number and name at the terminating set on an individual call basis.

From a mobile telephone, the Mobile X user first dials the Mobile X Service DN, then the CPP FFC to prevent their telephone number and name from being displayed on a receiving telephone across the PSTN for the following external call. The user then continues by dialing the desired DN. Internal calls within the CS 1000 system have originating numbers or names displayed, even though the originating call has requested privacy.

Camp-On

When an Attendant extends a call to the Mobile user and the user is busy on the mobile telephone, the external call is camped-on to the user's mobile telephone. With WTA class of service on the mobile UEXT, a warning tone is provided to the mobile telephone when there is a call camped-on to the mobile telephone. If the user frees the mobile telephone within a specified time, the camped-on call rings the mobile telephone automatically. If not, the call returns to the Attendant as a recall.

Cellular Voice Mail Avoidance

In Release 5.5 and later, cellular Voice Mail (VM) must be disabled since the calls to mobileX users through CS1000 will be immediatelly answered by cellular VM in case mobile phone is turned off or out of range. Cellular VM avoidance feature in Release 7.0 allows mobileX user to keep cellular VM configured together with CS1000 Call Pilot VM. With proper configuration on CS1000, cellular VM is ignored for all calls to mobileX user through CS1000 whenever mobile phone is turned off or out of range. For those calls, destination desktop set will be ringing while cellular call leg answer will be ignored for predetermined amount of time (MBXT timer value) configured on CS1000. It is administrator's responsiability to set up this MBXT value properly. MBXT is set to 0 by default and it indicates that cellular VM is disabled.

Dialable Cell CLID

The Dialable Cell Calling Line ID (CLID) feature displays the CLID of both internal and external calls originated by mobileX user. The User can specify whether the CLID is to be displayed in geographic CLID (GEO CLID) mode or cellular CLID (CELL CLID) mode. In GEO CLID mode the desk phone CLID appears on the devices, and in CELL CLID mode, mobile phone CLID appears on the devices. A new Class of Service (CLS), Cell CLID Mode Allowed (CCMA) and Cell CLID Mode Denied (CCMD) are added on the UEXT for this feature to work.

The CLID is available in a format which the user can use to return the calls from the logs without modifying dial strings. This Dialable Cell CLID feaure is supported on the devices like desktop sets, Mobile phones, and OC clients. During configuration, the user can be set up as a GEO CLID user or CELL CLID user. Redialable CLID is achieved using the DAPC feature. The MBXOT prompt defines the outgoing CLID types through this route.

Redialable CLID

Redialable CLID is the addition to the Single Number Mobile Number Cellular CLID enhancements. Redialable CLID provides CLID to each supported user device in a format that allows the user to return the call from call logs without modifying the dial string.

In order to make calling CLID re-dialable, DAPC feature is used:

- For calls from OCS and MobileX set to a local destination DN, DAPC feature must be configured to add appropriate AC1/2 prefix to the calling CLID.
- For calls originated on MobileX user desktop set to local destination DN, hard-coded DAPC table0 is used appropriately.

- For calls originated by mobile user (any of 3 devices) to external enterprise destination DN, the DAPC must be configured on a destination CS1000. This will add appropriate prefix to calling CLID to facilitate call back without editing the CLID
- For calls originated by mobile user (any of 3 devices) to external PSTN destination DN, the E164 calling CLID will be provided, so the destination party can return the call.

Dialable Single Number Mobile Number

You can use the Dialable Single Number Mobile Number (SNMN) feature to reach a Mobile User either by PBX or a cell number. Dialable SNMN is an extension of Mobile Number Single Number feature. In the Mobile Number Single Number feature the caller can reach a MobileX user by dialing E.164 mobile number. With the Dialable SNMN feature, the CS 1000 detects mobile numbers based on the SPN configuration. CS 1000 supports this feature regardless of the device used to make the call. The Mobile Service Provider (MSP) must route all calls to the Call Server first, where the called number is the MobileX user cell number. The Call Server detects the mobile number received from the MSP and replaces the dialed number with the internal UEXT DN.

Limitations

Following are the limitations of this feature:

- After a transfer or conference, the second call MobileX user must dial access code 1 (AC1) before 0 and the E.164 number of a target DN.
- The MSP must not remove the leading 0 from the dialed mobile phone number. This is because if an external non-FMC dials a moblile phone number for example, 07xxxxx, and the MSP sends this call to CS 1000 as 7xxxxx, then no internal extension can start with 7.
- SNMN does not work if you enable Overlap signaling .
- FLEN should not be equal to 0 and it should be equal to all dialed digits after AC.
- MBXX can be multi-digit and must be the dialed digits followed after SPN.
- Universal EXTension ID should be equal to the mobile number without AC and SPN. Number for HOT P should not use the same SPN. This avoids the call recursion and allows the call to direct to the mobile phone.
- AC1 or AC2 must not be equal to 0.

Assumptions

This feature makes the following assumptions:

- Mobile numbers must start with a known prefix.
- The MSP must route all calls to CS 1000 first, where the called number is MobileX mobile number.
- CS 1000 routes all the internal calls directly to the called party.
- The dialed number must be in AC1 + 07xxxxxxxxx format and must be dialed from desktop phone, OCS client, or local internal sets. Where AC1 can be one of the following:
    - dialed number from a mobile set

- number inserted by an MSP
- added on CS 1000 using ISDN or ESN digit manipulation. For example, Internal Digit Conversion (IDC).

• Mobile number prefix is configured as SPN or MOBXX as 0 or 7 for example.

• To enable plug-in 27 PRI configuration is required.

• AC2 must be used only for private internal calls, where as AC1 is used to call local, national, international and SPN numbers.

• SPN or MOBXX (0 or 7) is also activated for calls where dialed number is in the format 01628 43 xxxx (which is local to CS 1000 where MobileX UEXTs are configured). 0 is detected, but the digit 1 does not match 7, so default route list index is used. To handle such calls, you must add a separate SPN from ISDN configuration.

Configuration

This section provides an example of the configuration for IDC and SPN.

Note:

The configuration for IDC is for ISDN or ESN and it is not a mobileX functionality.

Assume mobile numbers dialed from PSTN in UK as 07xxxxxxxxx

E.164 international mobile number: 44 7 xxxxxxxxx

E.164 national mobile number:: 7 xxxxxxxxx

Assume Mobile numbers dialed from PSTN in Germany as 015xxxxxxxx, 016xxxxxxxx, 017xxxxxxxx

Where, 0 is the national prefix to dial E.164 national numbers between different geographic areas in the same country.

Configure IDC as shown:

```
0044 -> 90
01  -> 901
02  -> 902
03  -> 903
04  -> 904
06  -> 906
07  -> 907
08  -> 908
09  -> 909
000  -> 9000
001->90001
002  -> 9002
003  -> 9003
005  -> 9005
006  -> 9006
007  -> 9007
008  -> 9008
009  -> 9009
0040  -> 90040
0041  -> 90041
0042  -> 90042
0043  -> 90043
0045-> 90045
0046  -> 90046
```

```
0047 -> 90047
0048 -> 90048
0049 -> 90049
```

Configure SPN under AC1 for UK as below:

```
SPN 0
RLI 34
SDRR MBXX
Type MBXX
MBXX 7
 …
MBXX 8
```

Use MBXX for MobileX only.

Group Hunt

The Mobile X DN can be a member of a Group Hunting Pilot DN. The mobile telephone can use the Group Hunt Deactivate FFC to prevent Group Hunt termination on the Mobile X DN. The FFC can be dialed after dialing the Mobile Service DN.

Intercept Computer

The Intercept Computer feature allows the system to use an intercept (Attendant assistance service) computer for storing and retrieving call messages. A call to an absent user's DN using the Intercept Computer feature is routed to a designated Intercept Position (ICP) DN.

A Mobile X user can activate/ deactivate the Intercept Computer feature from a mobile telephone, by dialing the Mobile Service DN to setup an ISDN trunk to the CS 1000 system, and then dialing the ICP FFC.

Remote Call Forward Activation and Deactivation

The Remote Call Forward feature can be used to activate and de-activate the Call Forward All Calls feature of a user’s UEXT. The Remote Call Forward feature is accessible from Attendant consoles (the RCFW key), internal stations (using the Remote Call Forward FFC), and from the PSTN by making a DISA call (followed by the RCFW FFC).

If a Mobile user does not have a desktop telephone, Call Forward All Calls can only be activated using the CFW FFC, or using the Remote Call Forward feature from a mobile telephone where the CLID does not match the Mobile DN in the CS 1000 system. Once activated, any calls to the user’s enterprise DN will be forwarded, instead of ringing the mobile telephone.

Mobile X Activation and Deactivation

The basic operation of a Universal Extension for receiving calls is based on the operation of the Personal Call Assistant. Therefore PCA activation and deactivation must be used to control on a per user basis whether calls to the user's enterprise DN are redirected to the user's mobile telephone. If the user's UEXT is deactivated using the PCAD FFC, calls to the user's DN does not ring the user's mobile telephone. After the user's UEXT reactivates, calls to the user's DN rings the mobile telephone again.

There are FFC codes for PCA Activation and Deactivation. This allows the user to activate and deactivate the PCA feature for an office extension by making a call using the Service DN or by making a DISA call from the PSTN.

If the Mobile X user dials the Service DN and the PCA FFC code to activate and deactivate the PCA feature for its own Mobile X DN, authentication of SCPW is not required.

Important:

PCA deactivation does not impact calls from the user's mobile telephone. A user can make calls from the mobile telephone even if PCA is deactivated.

You can activate and deactivate PCA for mobile-X feature using the following methods.

- DISA DN—After you dial the DISA DN and input the authentication code, if required, dial PCAA/PCAD FFC + DN of the desktop phone + SCPW + #. A confirmation tone sounds.

  Any user can activate or deactivate PCA for any DN if they know their SCPW, no matter what set type they use.
- MSA DN—After you dial the MSA DN, and input the authentication code, if required, dial PCAA/PCAD + #. A confirmation tone sounds.

  MSA DN can only be used from the user's mobile phone. A user can activate or deactivate the PCA for UEXT only for where their mobile phone associates.
- Local desktop phone—After you dial the DISA DN and input the authentication code, if required, dial PCAA/PCAD FFC + DN of the desktop phone + SCPW + #. A confirmation tone sounds.

  Any user can activate or deactivate PCA for any DN if they know their SCPW, no matter what set type they use. Incase of multiple UEXT, it's important to maintain this SCPW unique to each UEXT. This will facilitate specific PCAA/PCAD being invoked.

Hot P key information (not the DN String) in the UEXT TN changes to block the call ringing to cell phone only. For instance:

- After you deactivate the PCA feature, any calls to the shared DN do not ring to the mobile phone associated with the desk phone.
- After you activate the PCA feature, calls to the shared DN ring to the mobile phone associated with the desk phone.

Caution:

There is a potential inconsistency between the target DN for receiving calls and the Mobile X CLID (UXID) for originating calls. Mobile X users are responsible for knowing the correct sequence to restore the operation of incoming calls back to the Mobile X telephone. For example, Mobile X is programmed with the HOT P key as HOT P 13 7036139675374 under normal circumstances. The Mobile X user decides to change the target DN to 96139675081. All calls to the Mobile X DN will now call 96139675081. It is the Mobile X user's responsibility to know that there is a special prefix used to direct calls back to the Mobile X telephone so that it is not restored to 96139675374.

Ring Again No Answer

The mobile user activates the Ring Again (RGA) feature when the destination party is ringing and does not answer. The mobile user who activates the RGA no answer feature is notified when the next call at the destination party is terminated. The notification is a short call to the mobile number by CS 1000. That call is terminated as soon as the mobile set starts to ring. Mobile user will have one call in the missed calls list with the CLID of destination party indicating that the destination party is idle.

Following is the description of how the RGA no answer feature works:

1. Mobile user dials MSA DN and then destination party DN. Call is in ringing state
2. Mobile user presses MFAC and then RGAA FFC, and then disconnect the call.
3. Once the next call at the destination party is terminated, the mobile user is notified by a short call from CS1000. RGA is automatically disabled at this point.

The caller (mobileX user) cannot do RGAA on another call when RGAA is already active. The caller must then call MSA DN, press MFAC+RGAD FFC and disconnect. Then, the caller can make a new call and activate RGA.

For more information on the Ring Again No Answer feature, see *Features and Services Fundamentals, Volume 5, NN43001-106*.

Ring Again when Mobile telephone Busy

If a Mobile X user calls to a mobile user and the mobile telephone is busy, the Mobile X user can activate the Ring Again on the mobile user. When the mobile telephone becomes idle, the Mobile X user who activated the RGA feature is notified by the CS 1000 server. The CS 1000 calls the mobile user's cell phone and disconnects when the phone starts ringing. So, a missed call from the mobile telephone is logged on the Mobile X user's cell phone and this notifies that the mobile user is now idle.

For more information on the Ring Again when Mobile telephone Busy feature, see *Features and Services Fundamentals, Volume 5, NN43001-106*.

Mid-Call features with traditional User Interface

In traditional User Interface (UI), the Mid-Call features are activated by dialling MFAC, Mid-Call FFC codes and the target DN. The Mobile user, while activating the Mid-Call features must memorize the FFC codes and decide which FFC code (transfer, conference, toggle) he wants to use before dialing the target DN. Memorizing and deciding which FFC code to use prior to dialling the target DN poses a disadvantage for using this Mid-Call features with traditional UI.

Mid-Call features with simplified UI

The Mobile user can use the simplified UI for Mid-Call features easier than with traditional UI. In the simplified UI, the Mobile user is in an established call and initiates the second call using Mid-Call features by dialing MFAC and the target DN. An ordinary dial tone is played to the Mobile user after pressing MFAC and the first call is automatically placed on hold. After the Mobile user is on the second call and decides to use the Mid-Call features, the Mobile user dials MFAC and MPO control codes to conference, toggle, or cancel the second call. To complete a blind or consultation call transfer, the Mobile user needs only to disconnect the call while the second call is established or is in a ringing state.

The Simplified UI has four MPO control codes. These codes come from Programmable Control Digits in MPO data block configured in LD15. The other one is MFAC. Following is the description of various control codes used in the Mid-Call features.

MFAC—Mobile-X Feature Activated Code

If the Mobile user enters MFAC, the current active call is held and the mobile trunk receives a dial tone. The DTR of CS 1000 collects all digits dialed from mobile trunk and do DN translation. This code is configured in FFC data block and is used by both traditional and Simplified UI.

CNFD—Conference Digit

After the second call is established, this code moves the first and second calls into conference. If either the first or second call is not in an established state, the CNFD is ignored. After 2 seconds, call reverts to the second call (if still established) or first call (if still established). If both calls disconnect, mobile user disconnects as well.

TGLD—Toggle Digit

When the second call is established, this code switches the mobile phone speech from current active call to the other call, which is on hold. Currently, yopu cannot invoke toggle from an unestablished call. This limitation is also applied to traditional UI. In other words, the first call cannot be restored if second call is not answered. Any toggle attempt is ignored before the second call is answered.

DISD—Disconnect Digit (corresponding to cancel in traditional UI)

As with FFC MCAN, DISD disconnects the current active call. If current call is the second call, this code stops the ringing. After the current call disconnects, the previous call, which is on hold, is automatically restored.

Operating parameters

The Mobile Extensions ISM parameters must be provisioned for this feature.

Only one mobile telephone is supported per mobile user. Only one line is supported on the mobile telephone. A second (or more) UEXT can be configured for the same DN, but each Mobile User must have a unique UEXT assigned to it.

The CS 1000 imposes a limit of 12 000 mobile extensions for each customer.

The CS 1000 Mobile Extensions feature is only supported for CPPIV and CP PM. It is not supported on Small System platforms that are based on the SSC Call Processor.

One Universal Extension is configured for each mobile telephone.

When a call is made from the mobile telephone, the CLID in the setup message must be consistent with the UXID configured in the Mobile X Universal Extension for that Mobile user, otherwise the call will be blocked.

If a Mobile user is equipped with a desktop telephone, this extension must share the DN with the prime key of the user’s Universal Extension TN in a Single Call Ringing (SCR) or Multiple Call Ringing (MCR) arrangement. Either the desktop TN or UEXT TN can be configured as

Multiple Appearance Redirection Prime (MARP). The MARP TN is used to determine how a call is redirected.

The full Mobile telephone DN (including area code), programmed as the UXID, must have no more than 16 digits.

A Universal Extension Mobile X set will be configured for each Mobile user with a Hot P key programmed with dialing information to reach the user's mobile telephone. If the HOT P key information is changed by using FFC PCAA, the UXID information of the UEXT TN is not affected. There will be a potential inconsistency in the configuration of DN to extend call to the mobile telephone and the CLID of incoming call from the mobile telephone.

Source based routing is the preferred interface for the mobile operator to route a call from a mobile telephone to CS 1000 system. If source based routing is not available, the Mobile user must dial the Mobile Service DN first to gain access to the CS 1000 system. ISDN PRI routes are used to connect to the Mobile Network. When source based routing is used, all calls made on the mobile telephone are routed by the Mobile Network to the CS 1000 Call Server where the Mobile X user data is configured.

A call will no longer be monitored for the MFAC if digits other than the MFAC are detected. This permits the user to make use of end to end signaling (EES) to provide information to an application without the risk of the software detecting the MFAC in the middle of a sequence and providing dial tone. The monitor will remain off for the remaining duration of the call. This means that once digits other than MFAC are dialed a call transfer or conference is not possible.

Mobile calls that are established when the system initializes are rebuilt as any other call on a PRI trunk. DTMF monitoring is not enabled for these calls after INI, so transfer and conference cannot be initiated by the mobile user. The status of the UEXT associated with the mobile user will not track the busy status of the mobile user until the next telephone call is made.

DPNSS interface and D70 ISDN interface are not currently supported.

The UXID of a UEXT only supports one CLID format. If the UXID is programmed with a CLID in National format and a call is received with the CLID in International format the call will be handled as a call from an unknown CLID.

Ring Again from the mobile telephone is not supported.

SIP and H.323 trunks cannot be monitored to access FFC features in the Call Server. If a SIP or H.323 trunk accesses the MSA DN the call will not have access to transfer and conference capabilities.

Release 7.0 supports MobileX feature over a SIP trunk. All Mid-Call features supported over PRI work the same way over SIP. To use mid call features over SIP, you must deploy the SIPX server between Home CS 1000 and the mobile service provider. SIPX server anchors media for all the calls between CS 1000 and the mobile service provider. Although the MobileX feature works over SIP trunk, this feature is not supported over an H.323 trunk.

If the mobile telephone call is routed to a CS 1000 Call Server which does not have the Mobile X user data configured (i.e., not the home PBX for the Mobile X user), the call is blocked.

The Mobile Network is expected to handle emergency and special service calls. These calls are not expected to be routed to the CS 1000 Call Server.

Calls made to the mobile telephone or from the mobile telephone that are not routed through the CS 1000 Call Server would not be known to the CS 1000 Call Server. As such, the CS 1000 Call Server would not be able to track the busy status of the mobile telephone.

While the mobile telephone is connected to another party within the Enterprise Network, the Mobile X user must press the special key sequence (MFAC, for example “*”) to trigger the CS 1000 system to initiate the Call in Progress features.

The mobile telephone must support DTMF tone generation when the user is pressing a key on the mobile telephone. A special dial tone will be provided to the mobile telephone when the MFAC is detected by the system.

Each call made to or from the mobile telephone consumes an ISDN trunk.

Each Mobile user requires a Universal Extension TN.

The FLEN parameter associated to a given MobileX PRI route should be less than or equal to the HOT P number length for MobileX UEXT. Otherwise, there will be a CS1000 delay in mobile extension ringing until the "End of Dialing" timeout is reached.

Mobile X users do not receive Alternate Call Routing treatment for unregistered devices while the Secondary Call Server is in the active state.

Feature packaging

The following packages must be provisioned to activate this feature:

- Mobile X (MOBX) package 412 – for configuring Mobile X UEXT for mobile users
- Personal Call Assistant (PCA) package 398 – for programming dialing information to reach the user on the mobile telephone and for basic operations of the Universal Extension
- ISDN package 145 and associated ISDN/Networking packages – for configuring the Mobile X network routes

Feature implementation

There are two possible configurations of the Mobile Extensions trunk Routing:

- Source based routing (With a dedicated mobile route to the PBX)
- Mobile Service Access DN (Without a dedicated mobile route to the PBX)

The implementation of the Mobile Extension Trunk Routing varies depending on the configuration used. The following task summary list table shows which procedures need to be followed for each option.

Caution:

There is potential impact on service interruption during the configuration of the Mobile Extensions Trunk Routing. It is recommended that the person implementing the feature should be familiar with the overlays used and that the implementation should be done during a maintenance window.

Task summary list

	Source based routing Implementation	Mobile Service Access DN Implementation		
Task	**Using CS 1000 Overlays (CLI)**	**Using CS 1000 Element Manager (GUI)**	**Using CS 1000 Overlays (CLI)**	**Using CS 1000 Element Manager (GUI)**
1.	Table 203: LD 15—Enable PCA at the Customer Level on page 605		Table 203: LD 15—Enable PCA at the Customer Level on page 605	
2.	Table 204: LD 15—Configure MFAC for Call in progress on page 605		Table 204: LD 15—Configure MFAC for Call in progress on page 605	
3.	Table 205: LD 15—Configure ringing cycles for Call Forward No Answer on page 605		Table 205: LD 15—Configure ringing cycles for Call Forward No Answer on page 605	
4.	Table 206: LD 22—Check equipped status of the Mobile X package on page 605		Table 206: LD 22—Check equipped status of the Mobile X package on page 605	
5.	Table 207: LD 16—Configure a Dedicated Mobile Route on page 606	Configuring a new Mobile Route on page 612	N/A	N/A

	Source based routing Implementation	Mobile Service Access DN Implementation		
Task	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)
6.	N/A	N/A	Table 208: LD 24-—Configure a Mobile Service DN with Security Code on page 607	Adding a new Mobile Service Directory Number on page 614, Editing an existing Mobile Service Directory Number on page 615, Deleting Mobile Service Directory Number on page 616
7.	Table 209: LD 20—Print MOBX data on page 607		Table 209: LD 20—Print MOBX data on page 607	
8.	Table 210: LD 20—Print UEXT TN block on page 607		Table 210: LD 20—Print UEXT TN block on page 607	
9.	Table 211: LD 11—Configure Mobile X users on page 608	Basic Client Configuration (BCC) on page 626, Configuring Features and Keys on page 627, Copying Class of Services from Desktop telephone to UEXT-MOBX on page 629, Configuring Features and Keys on page 631	Table 211: LD 11—Configure Mobile X users on page 608	Basic Client Configuration (BCC) on page 626, Configuring Features and Keys on page 627, Figure 62: Copy Class of Service from desktop telephone on page 631, Configuring Features and Keys on page 631
10.	Table 212: LD 11—Configure a Handoff key for a		Table 212: LD 11—Configure a Handoff key for a	

	Source based routing Implementation	Mobile Service Access DN Implementation		
Task	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)
	Mobile X desktop phone on page 608		Mobile X desktop phone on page 608	
11.	Table 213: LD 57—Configure Flexible Feature Codes for Mobile Call In Progress Features on page 609	Flexible Feature Code Entries page on page 618, Searching for Flexible Feature Codes on page 618, Searching for Flexible Feature Codes by Value on page 619, Searching for Flexible Feature Codes (Advanced) on page 620, Configuring Flexible Feature Codes (Basic) on page 620, Configuring Flexible Feature Codes (Advanced) on page 620, Editing FFC codes on page 621, Editing Group Call Code on page 622	Table 213: LD 57—Configure Flexible Feature Codes for Mobile Call In Progress Features on page 609	Flexible Feature Code Entries page on page 618, Searching for Flexible Feature Codes on page 618, Searching for Flexible Feature Codes by Value on page 619, Searching for Flexible Feature Codes (Advanced) on page 620, Configuring Flexible Feature Codes (Basic) on page 620, Configuring Flexible Feature Codes (Advanced) on page 620, Editing FFC codes on page 621, Editing Group Call Code on page 622

LD tables

The following tables provide Mobile Extensions implementation prompts and responses.

Table 203: LD 15—Enable PCA at the Customer Level

Prompt	Response	Description
REQ:	CHG NEW	Change or create new.
TYPE	FTR_DATA	Features and Options Data Block.
PCA	ON	Enable Personal Call Assistant.

Table 204: LD 15—Configure MFAC for Call in progress

Prompt	Response	Description
REQ:	CHG	Change.
TYPE	FFC_DATA	Flexible Feature Codes Data Block.
...		
CUST	0-99	Customer Number as defined in LD 15.
MFAC	x	Mobile Extension Feature Activation Code.

Table 205: LD 15—Configure ringing cycles for Call Forward No Answer

Prompt	Response	Description
REQ:	CHG	Change.
TYPE	RDR_DATA	Call Redirection data block.
CUST	0-99	Customer Number as defined in LD 15.
CFTA	YES	Call Forward to Trunk Access code allowed.
CFNA	1-(4)-15	Number of normal ringing cycles for Call Forward No Answer. It is recommended that you configure CFNA to a minimum of 4 rings. To steer all unanswered calls to the desktop mail box, configure the mobile device Call Forward No Answer to one higher than the desktop Call Forward No Answer.

Table 206: LD 22—Check equipped status of the Mobile X package

Prompt	Response	Description
REQ	PRT	Print.
TYPE	PKG 412	List the equipped packages.

Table 207: LD 16—Configure a Dedicated Mobile Route

Prompt	Response	Description
REQ:	CHG NEW	Change or create a new route.
TYPE:	RDB	Type of data block = RDB (Route data block).
CUST	xx	Customer number as defined in LD 15.
ROUT	x?x	Route number.
...		
TKTP	TIE	Trunk Type.
...		
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI Packet handler route. (Only appears for when TIE trunk type is selected.)
DGTP	PRI	Digital Trunk Type for route.
IFC	a…a	Interface type for this PRI route. The trunk interface of this mobile route is supported on most of the ISDN interfaces including EURO, ESIG, QSIG, D100, and NI2.
...		
CNTY		Country code.
CLID	a…a	Calling Line ID.
ISDN	YES	Dedicated Integrated Services Digital Network (ISDN) route.
...		
MODE	PRA	Mode of operation.
IFC	EURO, ESIG, QSIG, D100, NI2	Interface type for this PRI route. The trunk interface of this mobile route is supported on most of the ISDN interfaces including EURO, ESIG, QSIG, D100, and NI2.
MBXR	YES	Enable route for Mobile Extension.
...		
SIND	YES	Screening Indicator for the Mobile Extension route.
ICOG	ICT or IAO	Incoming and Outgoing trunk.
ACOD	x…x	Access Code for the trunk route.

Table 208: LD 24—–Configure a Mobile Service DN with Security Code

Prompt	Response	Description
REQ:	CHG NEW PRT	Change route, create a new route, print Mobile Service DN information.
TYPE:	MSA	Mobile Service Access.
CUST	x...x	Customer number as defined in LD 15.
SPWD	x...x	Security Data Password.
DN	x...x	Mobile Service DN, the DN can be up to 7 digits.
SCOD	x...x	Security Code (1 to 8 digit DISA security access code).
AUTR	YES	Enable Authorization Code Required.
CCBA	YES	Enable Allow Collect Call Blocking Answer Signal to be sent.

Table 209: LD 20—Print MOBX data

Prompt	Response	Description
REQ	PRT	Print.
TYPE	MOBX	Print the Universal Extension data for a MOBX type UEXT.
UXID	x...x	Enter the Mobile DN that the user wishes to print (up to 16 digits in length). A UXID must be specified. There is no means to print all UXIDs. (That can be done by printing the UEXTs.)

Table 210: LD 20—Print UEXT TN block

Prompt	Response	Description
REQ	PRT	Print.
TYPE	TNB, UEXT	TN block, or Universal Extension.
TN:	l s c u	Enter TN that user wishes to print.
CUST:	0-99	Customer number as defined in LD 15.
UXTY	aaa	Prints the UEXT type, if the TN type is UEXT.
UXID	x...x	Prints the mobile DN if the TN is UEXT.

LD 86, 87, and 90 Configure the Access Code to reach the Mobile Network

To configure the Access Code to reach the Mobile Network see Software Input Output Administration (NN43001-611), Software Input Output Reference –Maintenance

(NN43001-711), and Electronic Switched Network Reference - Signaling and Transmission (NN43001-280).

Table 211: LD 11—Configure Mobile X users

Prompt	Response	Description
REQ:	CHG/NEW	Change or create a new mobile Universal Extension.
TYPE:	UEXT	Universal Extension: This parameter indicates that this is a universal extension unit.
TN	l s c u	Universal extension TN.
CUST	0-99	Customer number as defined in LD 15.
UXTY	MOBX	Mobile Extension.
SCPW	xxxx	Station Control Password. The Station Control password is used for the Electronic Lock and Remote Call Forward features. This entry must equal the Station Control Password Length (SCPL) as defined in LD 15. Not prompted if SCPL = 0. See Flexible Feature Codes in the Features and Services Fundamentals (NN43001-106) Book 3 of 6.
UXID	x...x	For MOBX, this is the Calling Line ID of the mobile telephone Client number. It can be up to 16 digits. It must be entered for MOBX but is optional for other UXTY values.
KEY	0 SCR xxxx	Prime DN can be up to 7 digits.
KEY	01 HOT P nn yyy zzzz	Access Code to dial the Mobile Extension client's mobile telephone. <nn> is the maximum number of digits for HOT P DN (maximum 32 digits) <yyy> is the access code to dial to the Mobile Network. (For example: TSC (Trunk Steering Code) <zzzz>is the mobile telephone DN.

Table 212: LD 11—Configure a Handoff key for a Mobile X desktop phone

Prompt	Response	Description
REQ:	CHG NEW	Change or create a new Mobile X user.
TYPE:	a...a	The Set Type must support feature keys .
TN	LSCU	Terminal Number.
KEY	xx HNDO	Select Hand-off for a Mobile Extension user.

Important:

Personal Call Assistant (PCA) configuration is required. To configure PCA, see Personal Call Assistant in the Features and Services Fundamentals (NN43001-106) Book 5 of 6.

Table 213: LD 57—Configure Flexible Feature Codes for Mobile Call In Progress Features

Prompt	Response	Description
REQ:	CHG	Change FFCs.
TYPE	FFC	Feature Flexible Code.
CUST	0-99	Customer number as defined in LD 15.
FFCT	YES	Flexible Feature Confirmation Code.
CODE	MTRN	Activate the Mobile Extension transfer feature.
MTRN	xxxx	Enter the FFC used for the Mobile Extension transfer feature.
CODE	MCFA	Activate a conference from a mobile telephone.
MCFA	xxxx	Enter the FFC used to activate a conference from a mobile telephone.
CODE	MCOM	Complete a conference or transfer from a mobile telephone.
MCOM	xxxx	Enter FFC used to complete a conference or transfer from a mobile telephone.
CODE	MCAN	Cancel a Transfer or Conference from a mobile telephone.
MCAN	xxxx	Enter FFC used to cancel a transfer or conference from a mobile telephone.
CODE	MTGL	Enables a mobile telephone use to toggle between the two parties in a conference or transfer.
MTGL	xxxx	Enter FFC to enable a mobile telephone user to toggle between the two parties in a conference or transfer.

Table 214: LD 16—Configure Cellular Voice Mail Avoidance

Prompt	Response	Description
REQ:	CHG	Change.
TYPE	RDB	Type of data block = RDB (Route data block).
CUST	0	Customer number as defined in LD 16.
ROUT	3	Route Number.
...		
MBXOT		Mobile X Outgoing Type.
MBXT	0 (0—8000)	Mobile X timer. MBXT=0 indicates that the Cellular VM avoidance is disabled.

Prompt	Response	Description
DSEL		

Table 215: LD 15—Configure Simplified UI for Mid-Call features

Prompt	Response	Description
REQ:	CHG	Change.
TYPE	FFC	Flexible Features Code.
TYPE	FFC_DATA	Flexible Features Code Data Block..
CUST	0	Customer number as defined in LD 15.
...		
MFAC		Mobile-X Feature Activated Code. (Other codes are Conference Digit (CNFD), Toggle Digit (TGLD), and Disconnect Digit (DISD)).
MXSI	YES <Default Value NO>	Mobile X Simplified Interface.

Table 216: LD 11—Enable CLID in UEXT

Prompt	Response	Description
REQ:	NEW/CHG/PRT	Request New, Change, or Print.
TYPE	UEXT	Universal Extension.
CUST	0	Customer number as defined in LD 11.
...		
CLS	CCMA (or CCMD)	Cell CLID Mode Allowed (or Denied).

Table 217: LD 16—Configure re-dialable CLID

Prompt	Response	Description
REQ:	NEW/CHG	Request New or Change route.
TYPE	RDB	Type of data block = RDB (Route data block).
CUST	0	Customer number as defined in LD 16.
ROUT	xx	Route Number.
...		
MBXR	YES	Mobile Extension Route.
SIND	NO	Screening Indicator for Mobile Extension route.
MBXOT	NPA (or INTL)	National or International Mobile CLID.

Table 218: LD 90—Configure Single Number Mobile Number

Prompt	Response	Description
REQ:	NEW	New Request.
CUST	0	Customer number as defined in LD 90.
FEAT	NET	Feature = NET
TRAN	AC1	Translator (AC1 or AC2).
TYPE	SPN	Type = SPN (Special Number Translation).
SPN	0	Special Number Translation.
FLEN	11	Flexible Length.
RLI	2	Route List Index.
SDRR	MBXX	Supplementary Digit Restriction or Recognition.
MBXX	7	Mobile X prefix.

Element Manager

Introduction

Element Manager (EM) Mobile Extensions configurations are supported on CP PIV and CP PM systems only and must adhere to the following prerequisites:

- All required packages must be equipped. See Feature packaging on page 601.
- All necessary customer configuration must be implemented for a route to be configured.
- A user must be logged in to Element Manager on a CP PIV or CP PM system using a valid account.

Element Manager (EM) supports the following configuration and maintenance aspects of the Mobile Extension feature:

- Overlay 16, 21 & 22 Prompts are used to identify if the route configured is a dedicated mobile route or not. Element Manager supports the prompts for this configuration along with the existing support for a route configuration.
- Overlay 24 A data block "Mobile Service Directory Number" is used to provide connectivity between mobile network and enterprise network. An MSDN can be dialed by a mobile user in a mobile network to gain access to the enterprise network. Element Manager supports the data block.
- Overlay 57 Flexible Feature Codes are used to provision in-call feature for mobile users. Cs 1000 Release 5.0 Element Manager does not support any configuration using overlay

57. Overlay 57 is supported in Element Manager for CS 1000 Release 5.5 to provide complete configuration of Flexible Feature Codes.

- Maintenance: Overlays 80 & 143 The command results of these overlays are modified for this feature. EM supports the change in output in these overlays by testing the response from these overlays.

  For more information, see *Software Input Output Reference — Maintenance, NN43001-711*.

Element Manager configuration

This section provides Element Manager configuration procedures for the Mobile Extensions feature.

Configuring a new Mobile Route

1. Navigate to the Route and Trunks page.
2. Click **Add Route** for any configured customer.
3. Select the Trunk Type for which Integrated Services Digital Network option can be configured (TIE, COT, DID, ISA, CBCT etc).
4. Select the Digital Trunk Route check box and click **PRI** or **PRI2** for the **Digital Trunk Type** sub prompt.
5. Select the **Integrated Services Digital Network** check box.

   The Check box Mobile Extension Route is displayed and is cleared by default.
6. If the **Mobile Extension Route** prompt is selected, the sub prompt **Screen Indicator** displays and the check box is cleared by default.
7. Enter the value for all mandatory fields, and click **Submit**.

   The Routes and Trunks page is loaded with the new route entry (see Figure 42: Routes and Trunks on page 613.

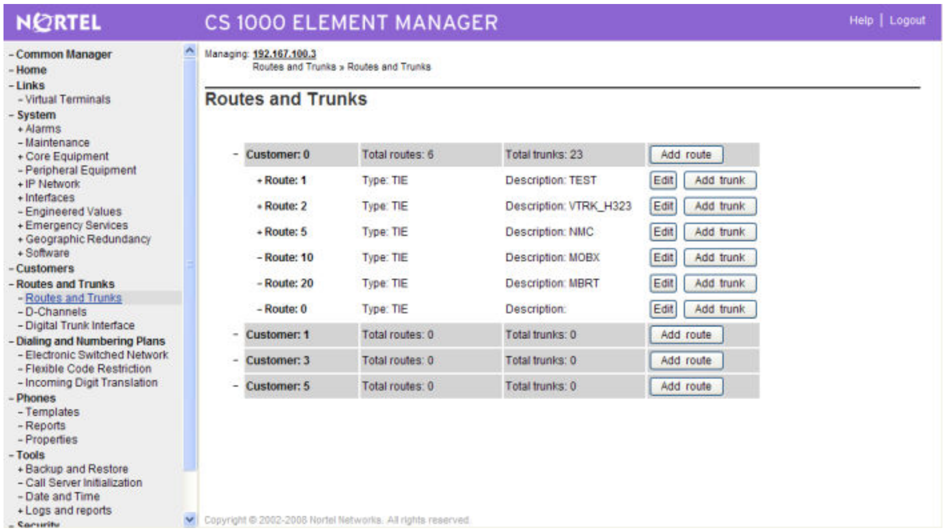


Figure 42: Routes and Trunks

Configuring the Mobile Service Directory Numbers page

1. Navigate to the Customers page from the navigation tree.
2. Click on # (customer number) hyperlink in the data grid.

   In the Edit page a new hyperlink Mobile Service Directory Numbers is displayed as indicated .
3.
4. Click on the hyperlink.

   A new page Mobile Service Directory Numbers displays the list of configured Mobile Service Directory Numbers in a datagrid, as indicated in Figure 44: Mobile Service Directory Numbers page on page 614.

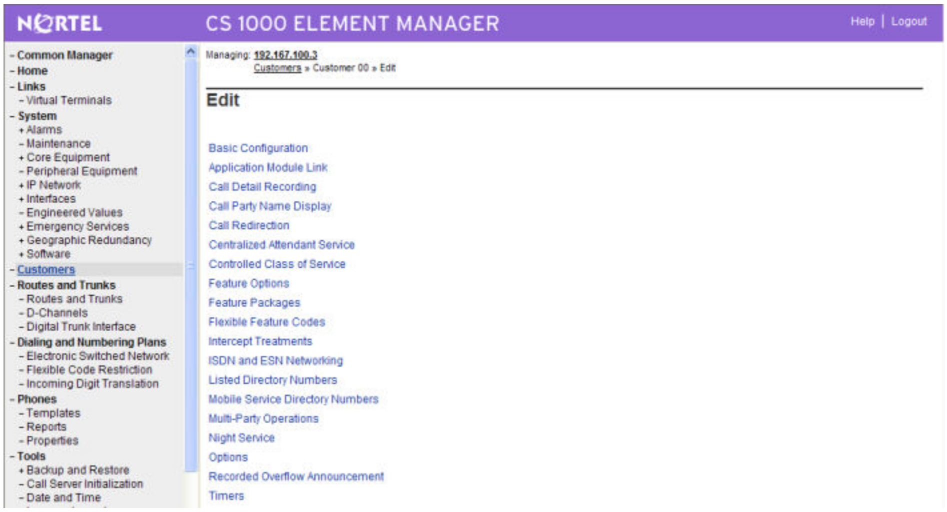


Figure 43: Mobile Service Directory Numbers link

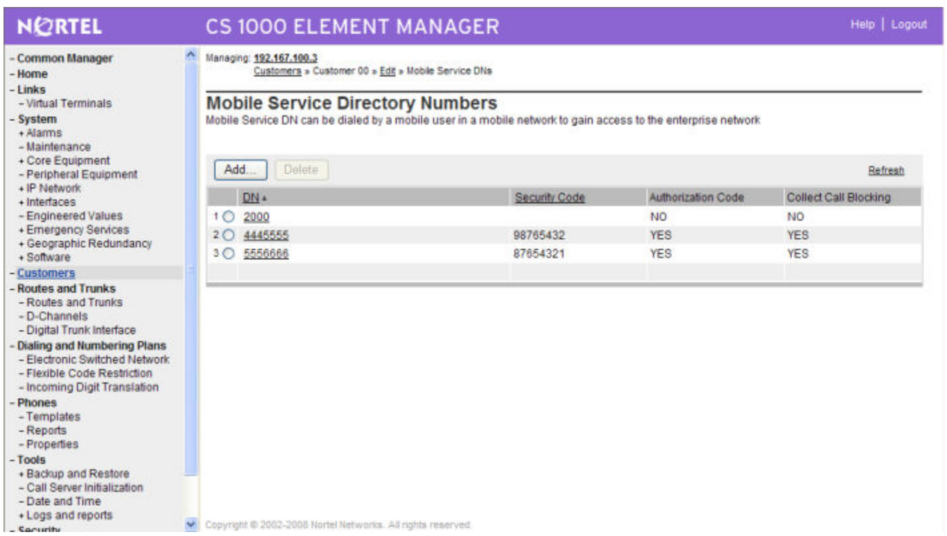


Figure 44: Mobile Service Directory Numbers page

Adding a new Mobile Service Directory Number

1. Navigate to Customers page from the navigation tree.
2. Click on # (customer number) hyperlink in the data grid.
3. In the Edit page click the Mobile Service Directory Numbers hyperlink to navigate to Mobile Service Directory Numbers page.
4. Click **Add** .

   The Add Mobile Service Directory Number page is displayed as indicated in Figure 45: Add Mobile Service Directory Numbers page on page 615.
5. Enter Mobile Service Directory Number (up to seven digits).
6. Enter Security code (up to eight digits).

7. Select the **Authorization Code Required** and **Collect Call Blocking Allowed** check boxes.
8. Click **Save**.

   The new Mobile Service Directory Number is configured and the Mobile Service Directory Numbers page is displayed.
9. Click **Cancel.**

   The Mobile Service Directory Numbers page is displayed.

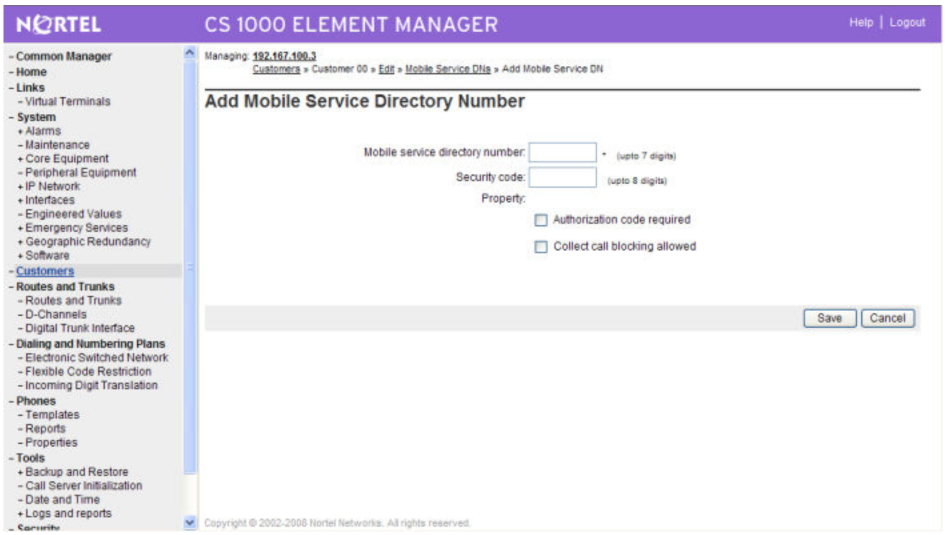


Figure 45: Add Mobile Service Directory Numbers page

Editing an existing Mobile Service Directory Number

1. Navigate to Customers page from the navigation tree.
2. Click on # (customer number) hyperlink in the data grid.
3. In the **Edit** page, click the Mobile Service Directory Numbers hyperlink to navigate to Mobile Service Directory Numbers page.
4. Click on # (Mobile Service Directory Number) hyperlink in the data grid.
5. The Edit Mobile Service Directory Number page is displayed as indicated in Figure 46: Edit Mobile Service Directory Numbers page on page 616 .
6. Click **Save**. The values edited for Mobile Service Directory Number are configured and the Mobile Service Directory Numbers page is displayed.
7. Click **Cancel**.

   The Mobile Service Directory Numbers page is displayed.

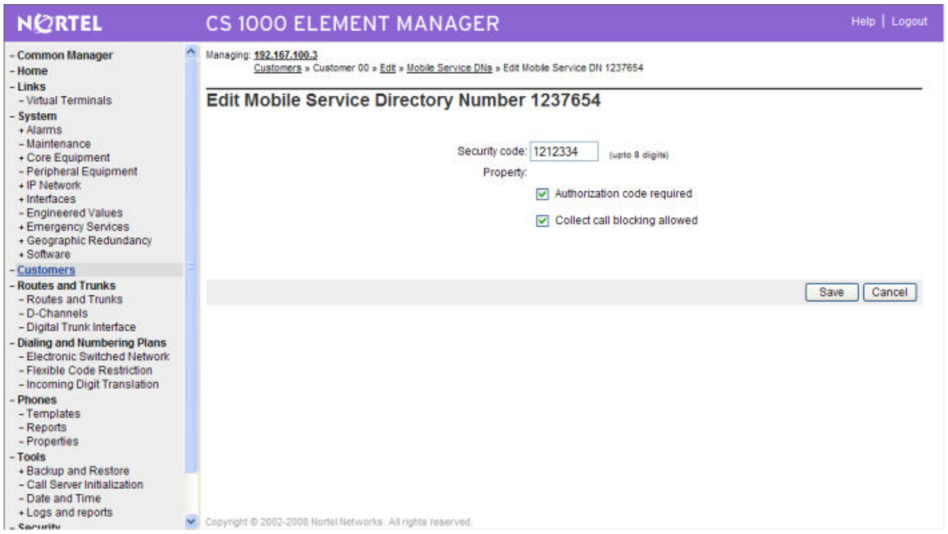


Figure 46: Edit Mobile Service Directory Numbers page

Deleting Mobile Service Directory Number

1. Navigate to Customers page from the navigation tree
2. Click on # (customer number) hyperlink in the data grid.
3. In the Edit page, select the Mobile Service Directory hyperlink to navigate to Mobile Service Directory Numbers page.
4. Click any of the radio buttons, and click **Delete**.
5. The selected entry is deleted and the page is reloaded.

Password protection for Mobile Service DN Configurations:

The Password window is displayed if the Secure Data Password is set in the password block for the Customer in LD 15.

After the password is entered, it is saved for future use. The Password window appears only for the first configuration of a session.

Adding a new Mobile Service Directory Number

1. Navigate to the Customers page from the navigation tree.
2. Click on # (customer number) hyperlink in the data grid.
3. In the Edit page, select the **Mobile Service Directory Numbers** hyperlink to navigate to Mobile Service Directory Numbers page.
4. Click **Add**.

   The Add Mobile Service Directory Number page is displayed.
5. Enter a Mobile Service Directory Number (up to seven digits).
6. Enter a Security code (up to eight digits).

7. Select the **Authorization Code Required** and **Collect Call Blocking Allowed** check boxes.
8. Click **Save**. The Password window is displayed as shown in Figure 47: Password window on page 617.
9. Enter the Secure Data Password and click **Save**. The window closes.
10. Click **Cancel** to close the Password window.
11. Click **Save** in the Add Mobile Service Directory Number page. The new Mobile Service Directory Number is configured and the Mobile Service Directory Numbers page is displayed. The configuration is successful only if the user enters the right password.
12. Click **Cancel**. The Mobile Service Directory Numbers page is displayed.

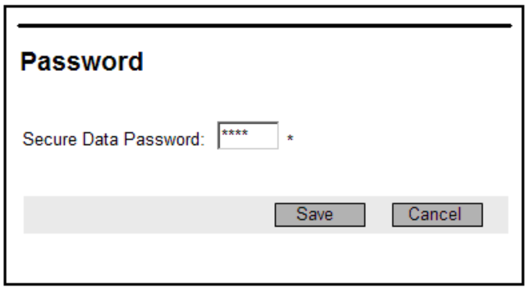


Figure 47: Password window

Deleting Mobile Service Directory Number

1. Navigate to the Customers page from the Navigator pane.
2. Click on # (customer number) hyperlink in the data grid.
3. In the Edit page, select the **Mobile Service Directory Number** hyperlink to navigate to the Mobile Service Directory Numbers page.
4. Click any of the radio button and click **Delete**.
5. Click **Delete**. The Password window is displayed.
6. Enter the Secure Data Password and click **Save**. The Password window closes.
7. The selected entries are deleted and the page is reloaded. The configuration is successful only if the user enters the correct password.

Flexible Feature Codes:

The Flexible Feature Codes Web page allows users to configure the Flexible Feature Codes (FFC) data block for a customer. Complete the following steps to navigate to the Flexible Feature Codes Web page:

1. Select Customers
2. Select the appropriate customer number from the list.
3. Select the Flexible Feature Codes link.

To configure Flexible Feature Code end of dialing indicator, select the **Provide end of dialing indicator** check box.

Enter the appropriate information, and click **Save**.

Flexible Feature Code Entries page

1. Navigate to **Customers, Select a customer number, Flexible Feature Codes** page.
2. The page should contain a Search block and a note is displayed below it indicating the user to Select your search criteria, enter or lookup the desired value and click Search. New flexible feature codes may also be added as indicated in Figure 48: Flexible Feature Code Entries page on page 618 .

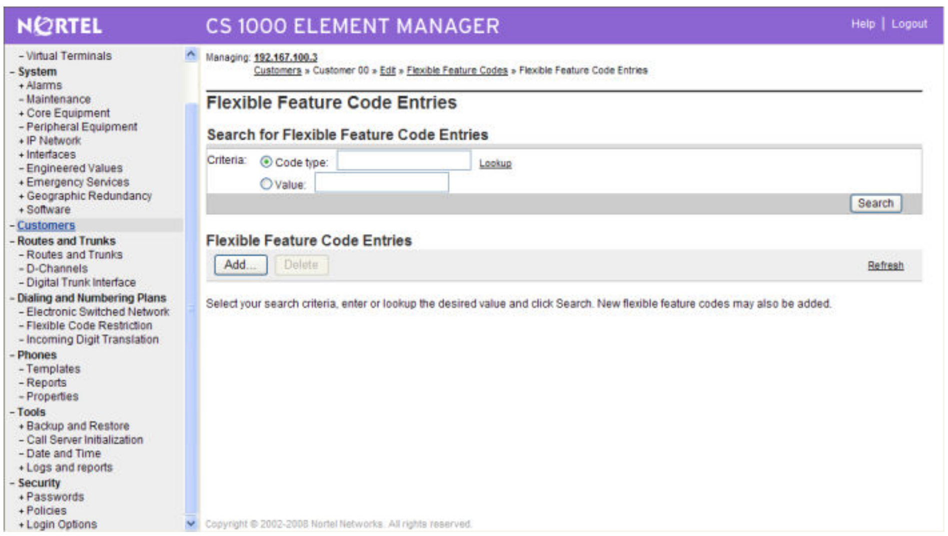


Figure 48: Flexible Feature Code Entries page

Searching for Flexible Feature Codes

1. Navigate to **Customers, Select a customer number, Flexible Feature Codes** page.
2. Click on the **Lookup** hyperlink in the Search block.

   The Flexible Feature Code Lookup window opens. See Figure 49: Flexible Feature Code Lookup on page 619.
3. Select the check box for the required features, and click **Assign**.
4. The selected feature prompt names should be displayed in the **Code type** input box.
5. Ensure that the **Code type** radio button is selected.
6. Click **Search** to display the codes for the selected feature in the datagrid. See Figure 49: Flexible Feature Code Lookup on page 619.

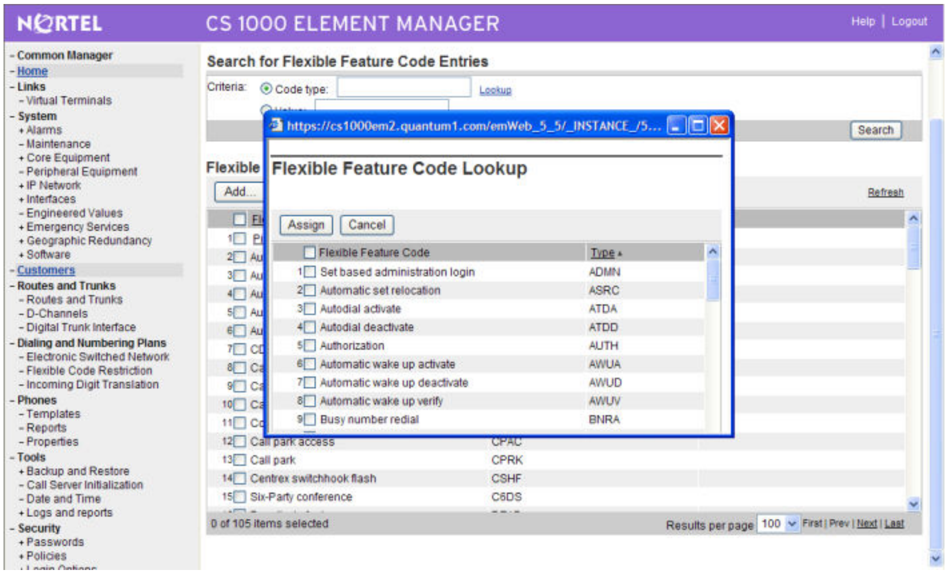


Figure 49: Flexible Feature Code Lookup

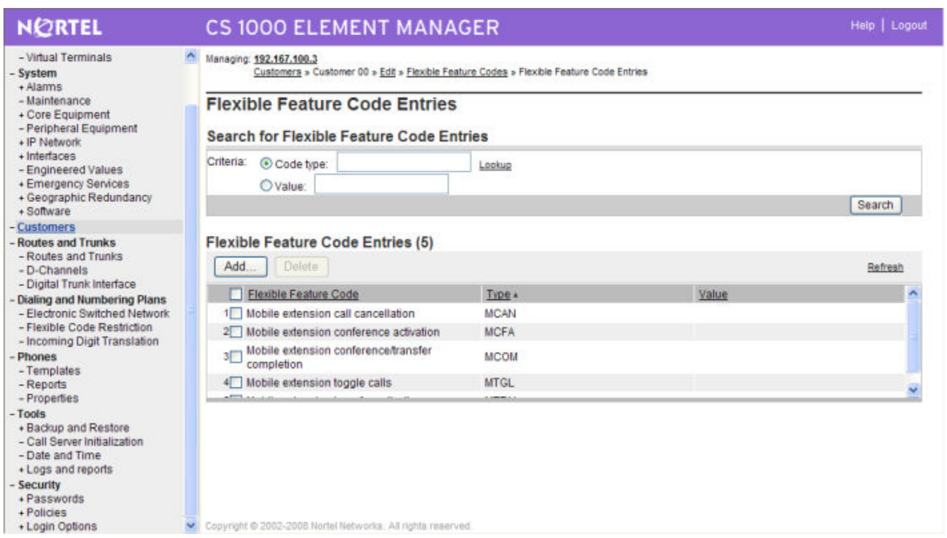


Figure 50: Flexible Feature Code Entries page after Search Operation

Searching for Flexible Feature Codes by Value

1. Enter the configured Flexible Feature Code value in the **Value** input box.
2. Select the **Value** radio button.
3. Click **Search**. The flexible feature code, type, and the value are listed in the datagrid.

Searching for Flexible Feature Codes (Advanced)

1. Navigate to **Customers, Select a customer number, Flexible Feature Codes** page.
2. Enter the wildcard character asterisk (*) before or after the search text in the **Code type** input box.
3. Select the **Code type** radio button.
4. Click **Search.** All the flexible feature code types with the configured values that match the given search text are listed in the datagrid.

Flexible Feature Code Configuration:

Element Manager does not have the capacity to configure more than 3 000 Flexible Feature Codes (i.e., PRT FFC ALL can result in Xmsg timeout). The maximum number of Flexible Feature Codes expected to be configured at one time is 300. There is no handling done for the paging of the Flexible Feature Codes.

Configuring Flexible Feature Codes (Basic)

1. Navigate to **Customers, Select a customer number, Flexible Feature Codes** page.
2. In the **Flexible Feature Codes** Web page, select the box **Provide confirmation tone**.
3. To complete the configuration, click **Save**.
4. Click **Cancel** to cancel the action.

   The Edit Web page is displayed.

Configuring Flexible Feature Codes (Advanced)

1. Navigate to **Customers, Select a customer number, Flexible Feature Codes** page.
2. Click **Add** in the Flexible Feature Code EntriesWeb page.

   The Add Flexible Feature Code Web page opens.
3. Click the **Lookup** adjacent to the **Flexible Feature Code type** input box.

   The Flexible Feature Code Lookup window opens. See Figure 51: Add Flexible Feature Code on page 621.
4. The Lookup lists all the Flexible Feature Code types and their corresponding descriptions.
5. Clicking on any of the code type hyperlinks selects the particular type into the **Flexible Feature Code type** input box. Also, you can enter FFC code type.
6. Enter a FFC in the range 0-####### in **Value** input box. Click **Save** to save the values.

Figure 51: Add Flexible Feature Code

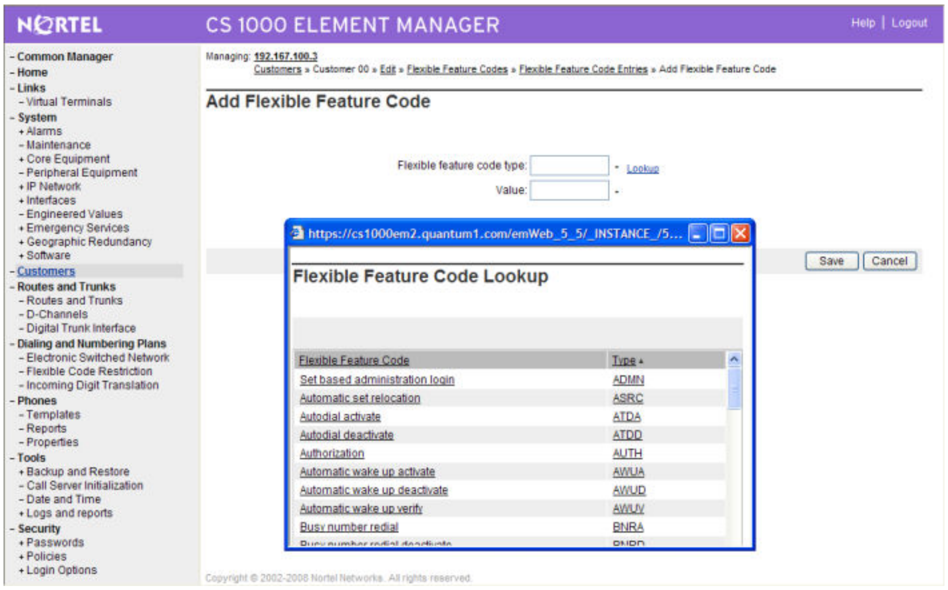


Figure 52: Add Flexible Feature Code Lookup

Editing FFC codes

1. Navigate to **Flexible Feature Code Entries** page.
2. The prompts configured for **Group call code, Pilot DN code** and **1xx special services** appear as hyperlink to navigate to the corresponding edit page.

Configuring Group Call Code

1. Navigate to **Customers, Select a customer number, Flexible Feature Codes, Flexible Feature Code Entries, Add Flexible Feature Code** page.
2. Select the **Lookup** hyperlink, and select **Group call code** from the lookup table.
3. The sub prompts **Group call list number** and **Value** that appear below the Flexible feature code type must be filled in as indicated in Figure 53: Add a Group Call code on page 622.
4. Click **Save**.

The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.

5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

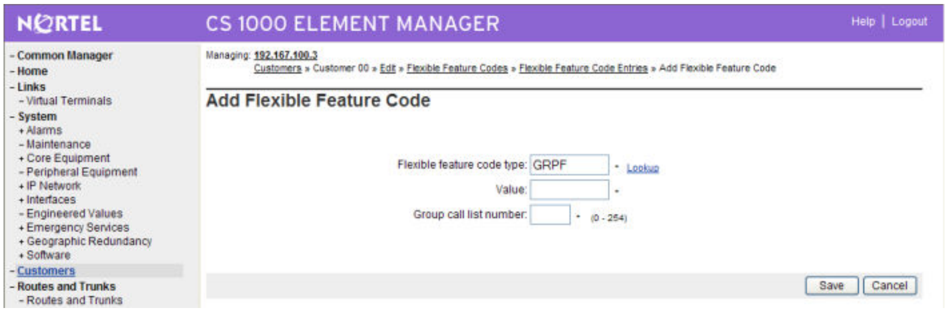


Figure 53: Add a Group Call code

Editing Group Call Code

1. Navigate to **Flexible Feature Code Entries** page.
2. Click on any of the configured # (GRPF) hyperlink.

   The Edit Group Call Code xxxxxxx page is displayed. See Figure 54: Edit Group Call code xxxx on page 622.
3. Edit the value.
4. Click **Save**.

   The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the edited entry.
5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

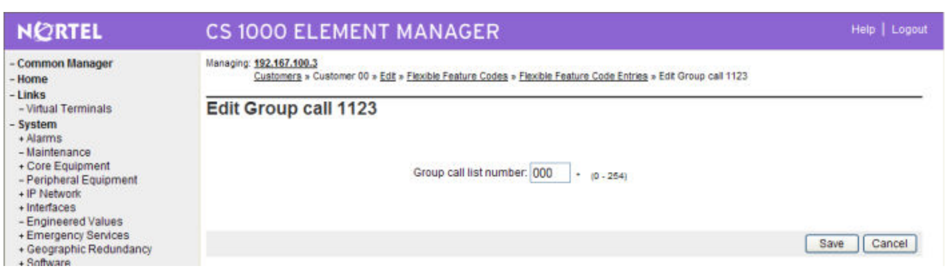


Figure 54: Edit Group Call code xxxx

Configuring 1xx special features code

1. Navigate to **Customers, Select a customer number, Flexible Feature Code, Flexible Feature Code Entries, Add Flexible Feature Code** page.
2. Click **Lookup**, and select ITXX from the lookup table.

3. The sub prompts CO route number for the '1xx' service and Value that appear below the Flexible feature code type must be filled in as indicated in Figure 55: Add 1xx special services code on page 623. Enter the values.
4. Click **Save**.

   The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.
5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

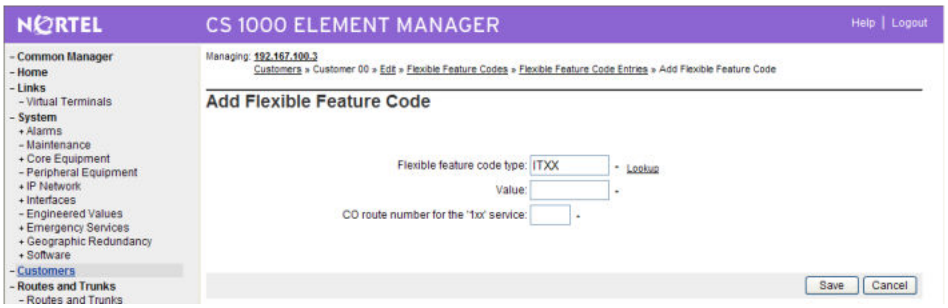


Figure 55: Add 1xx special services code

Editing 1xx special features code

1. Navigate to **Flexible Feature Code Entries** page.
2. Click on any of the configured ITXX hyperlinks.

   The **Edit For '1xx' Special Services xxxxxxx** for ITXX is displayed as indicated in Figure 56: Edit for 1xx special services code page on page 623.
3. Edit the value.
4. Click **Save**.

   The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the edited entry.
5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

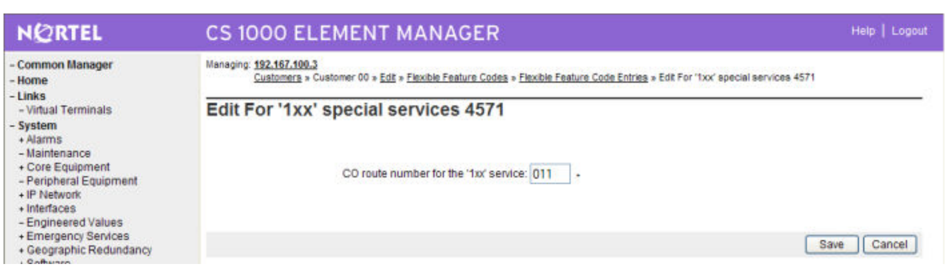


Figure 56: Edit for 1xx special services code page

Configuring Pilot Directory Number code – Use as Initiate Group Hunting

1. Navigate to the Add Flexible Feature Code page.
2. Select the **Lookup** hyperlink and select **PLDN** from the lookup table.

   The prompt List number and Use is displayed below the Value prompt.
3. The sub prompts of **Initiate group hunting, SCL/SSC list user, and SCL/SSC list controller** are displayed in the drop down menu of the **Use** field and are disabled by default as indicated in Figure 57: Add Pilot DN code on page 624.
4. Select **Use** and **Initiate Group Hunting**. The sub prompts get enabled. Enter the required values.
5. Click **Save**.

   The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.
6. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

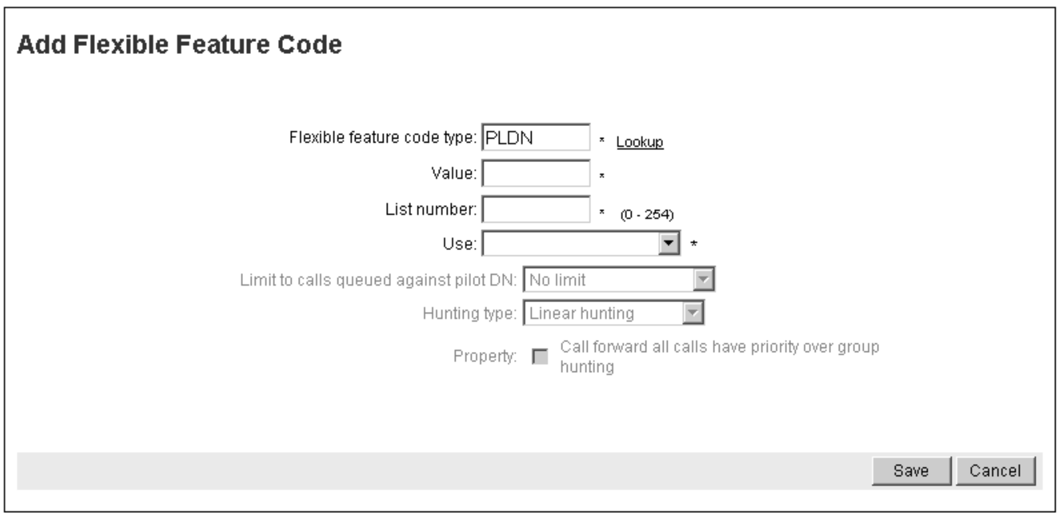


Figure 57: Add Pilot DN code

Configuring Pilot Directory Number code – Use as SCL/SSC List Controller or SCL/SSC List User

1. Navigate to Add Flexible Feature Code page.
2. Click on the **Lookup** hyperlink and select **PLDN** from the lookup table.
3. The prompt List number and Use is displayed below the Value prompt.
4. The sub prompts of **Initiate group hunting, SCL/SSC list user, and SCL/SSC list controller** are displayed in the drop down menu of the Use field and are disabled by default as indicated in Figure 57: Add Pilot DN code on page 624.
5. Select **Use** as **SCL/SSC list controller** or **SCL/SSC List User**. The sub prompts remain disabled. Enter the required values.

6. Click **Save**.

   The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.

7. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

Editing Pilot Directory Number code – Use as Initiate Group Hunting

1. Navigate to Add Flexible Feature Code page.
2. Select any of the configured PLDN hyperlinks.
3. The Edit Pilot DN xxxxxxx is displayed as indicated. The List number and Use prompt is displayed as text field. Edit the values. The sub prompts of Use appear, which can be edited. Edit the values.
4. Click **Save**. The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.
5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

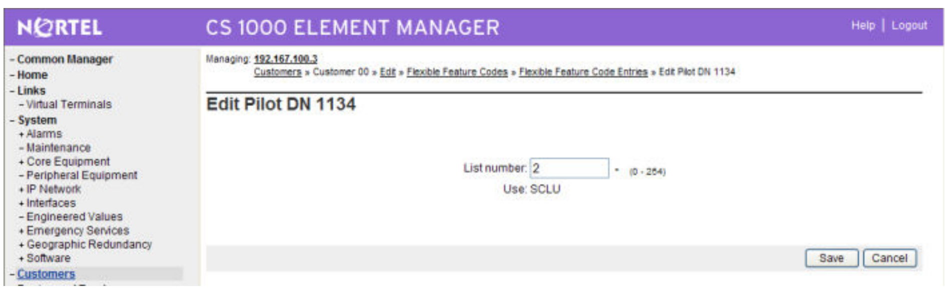


Figure 58: Edit Pilot DN code page – Use as Initiate Group Hunting

Editing Pilot Directory Number code – Use as SCL/SSC List Controller or SCL/SSC List User

1. Navigate to Add Flexible Feature Code page.
2. Click on any of the configured **PLDN** hyperlinks.
3. The Edit Pilot DN xxxxxxx is displayed as indicated in Figure 59: Edit Pilot DN code page - Use as SCL/SSC List Controller on page 626. Edit the values.
4. Click **Save**.

   The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.

5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

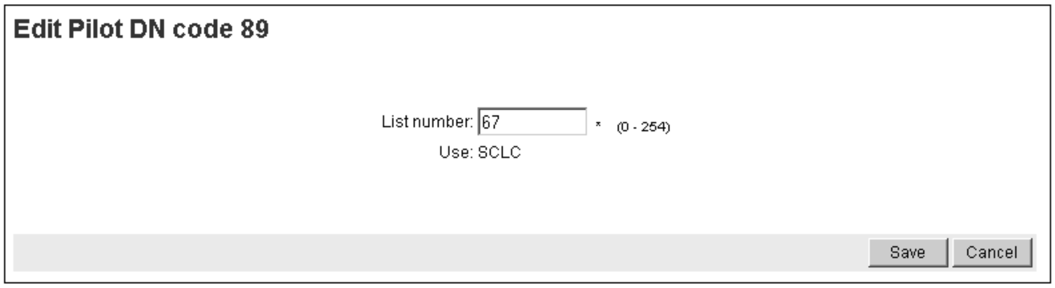


Figure 59: Edit Pilot DN code page - Use as SCL/SSC List Controller

Configuring Mobile codes

1. Navigate to Add Flexible Feature Code page.
2. Select any of the listed FFC: Mobile extension conference activation code, Mobile extension call cancellation code, Mobile Extension Conference/Transfer Completion code, Mobile extension toggle calls code or Mobile Extension Transfer Activation code from the lookup table.
3. Enter the appropriate value in the **Value** box.
4. Click **Save**. The Flexible Feature Code is configured and the Flexible Feature Code Entries page is displayed with the new entry.
5. Click **Cancel**.

   The Flexible Feature Code Entries page is displayed.

Mobile FFC – machine dependency

1. Navigate to Add Flexible Feature Code page.
2. The mobile specific FFC: Mobile extension conference activation code, Mobile extension call cancellation code, Mobile Extension Conference/Transfer Completion code, Mobile extension toggle calls code or Mobile Extension Transfer Activation code is absent for CP PII and small systems.

Error Scenario for Mobile Flexible Feature Codes

1. Navigate to **Add Flexible Feature Code** page.
2. Select any of the mobile feature flexible codes from the **lookup** table.
3. Enter a previously configured code for any of the mobile FFC (other than MFAC) and click **Save**.
4. The existing SCH error SCH8893 with description Specified DN conflicts with an existing DN is displayed.

Basic Client Configuration (BCC)

Introduction

Mobile Extensions introduces set type MOBX (Mobile Extension), FMCL (Fixed Mobility Converged Line), TLSV (Telephony Services), SIPN(SIP Line for Nortel Clients), and SIP3

(SIP Line for 3rd Party Clients) that provides a logical connection to the user's mobile telephone.

BCC supports the following functionality for UEXT sets:

- Single/bulk telephone addition
- Single/bulk telephone modification
- Basic and advanced search
- Retrieval of telephones from PBX
- Single/bulk telephone deletion
- Reports generation in HTML and CSV format
- Import operation

Prerequisites

BCC has the following prerequisites:

- This feature is supported in CP PIV and CP PM systems only.
- Swap and Move operations are not supported on these sets.
- While performing import operation on these sets, UXTY value will be ignored completely as UXTY value is deduced for set type name only.

Mobile X Manadatory Fields in BCC

UXID is mandatory for the UEXT-MOBX telephone type, and validation must be completed. If it is blank and errors generate,. KEY 0 must be configured as DN key (SCR, SCN, ACD, etc.) and KEY 1 must be configured as HOT P key.

Procedures

Refer to the following procedures to configure BCC.

UEXT-MOBX Configuration

Configuring Features and Keys

1. Logon to ECM using a valid account.
2. Click on the appropriate call server element. Element Manager is launched.
3. Click Phones link from the left navigation pane. The Search for Phones page appears.

    Note:

    telephone database and call server database must be kept synchronized all the time. Necessary configuration of a customer must already be available for a telephone set to be configured (superloop, customer etc).

4. From the Search for Phones page under the Phones heading, click Add. The New Phones page opens.
5. Select UEXT-MOBX-Universal Extension MOBX from the telephone Type drop down menu (see Figure 60: UEXT set configuration on page 628)

6. Enter the other applicable details under the Type and Options fields and click Preview. The telephone Details page appears. The top of the telephone Details page should show the telephone Type as Universal Extension MOBX.

   Note:

   Two features are added under the Features section:

   - UXTY (Universal Extension Type) for which the value is MOBX (Mobile X user). This field cannot be edited.
   - The prompt UXID (Universal Extension ID) is mandatory for Mobile X configuration. It can be up to 16 digits.

7. Enter the appropriate Universal Extension ID in the UXID field under the Features section.
8. Navigate to the Keys section of the telephone Details page. Key 0 is configured with the SCR DN. Key 1 is configured as a HOT_P key containing the mobile DN.
9. Enter the required details for Key 1 and Key 2 and click on the Validate button to perform validation of general properties, features and keys.
10. Click the Finish button.

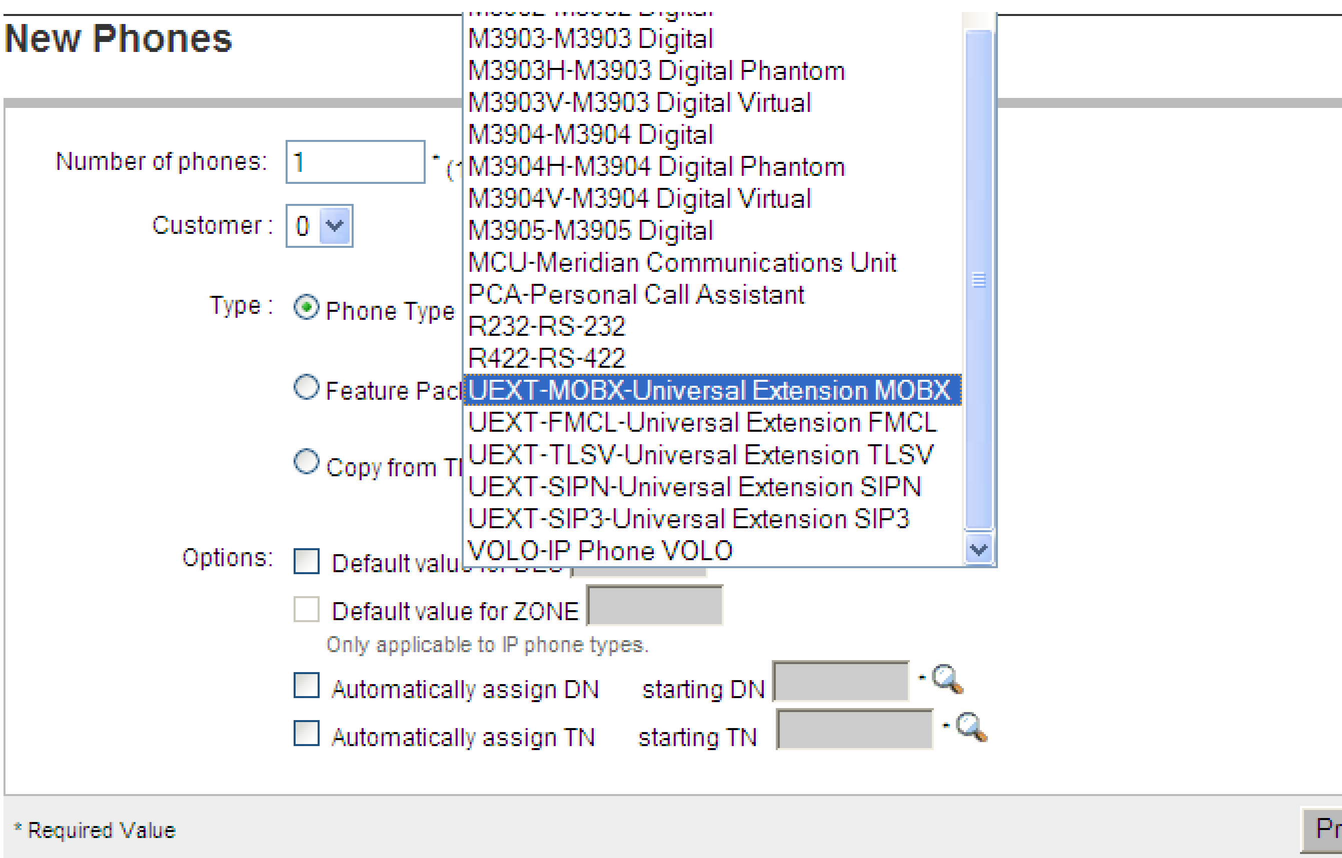


Figure 60: UEXT set configuration

Copying Class of Services from Desktop telephone to UEXT-MOBX

1. From the telephone Details page of the newly configured UEXT-MOBX telephone, in keys Section, key 0 is configured as any DN Key (SCR,SCN,ACD,MCN,MCR,PVN,PVR)
2. Select the appropriate Key Type from the list.
3. Enter DN value and click on the telephone handset image shown in Figure 61: Key configuration for UEXT sets on page 630.
4. A new page opens and then all desktop telephones that are already configured with the DN same as that entered in Key 0 of UEXT-MOBX telephone are listed in the page. See Figure 62: Copy Class of Service from desktop telephone on page 631.
5. If any one telephone is selected and Assign is clicked, the values of HUNT, FDN, CLS, NCOS & TGAR class of services values are copied from the telephone selected to the new telephone being configured.

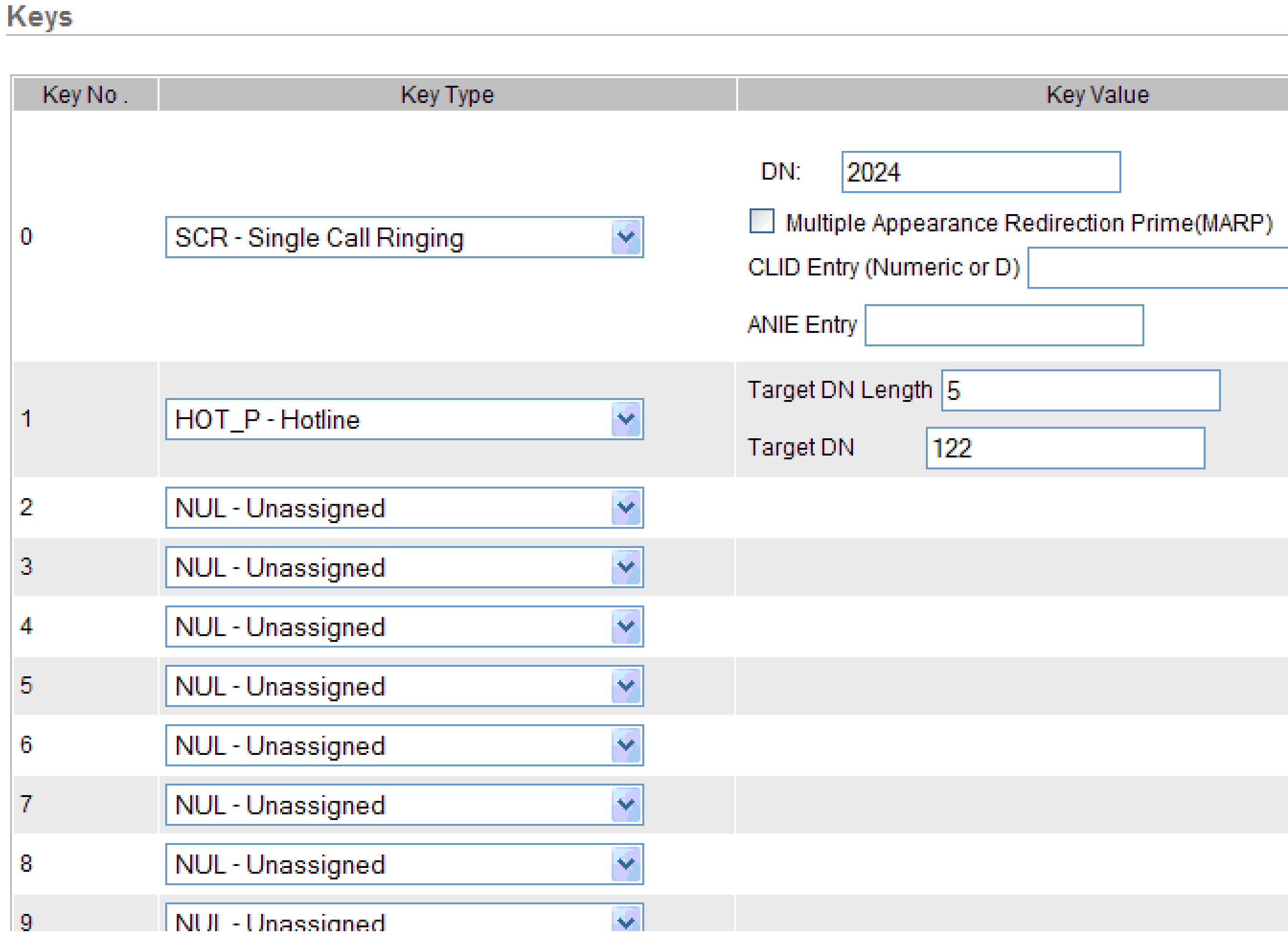


Figure 61: Key configuration for UEXT sets

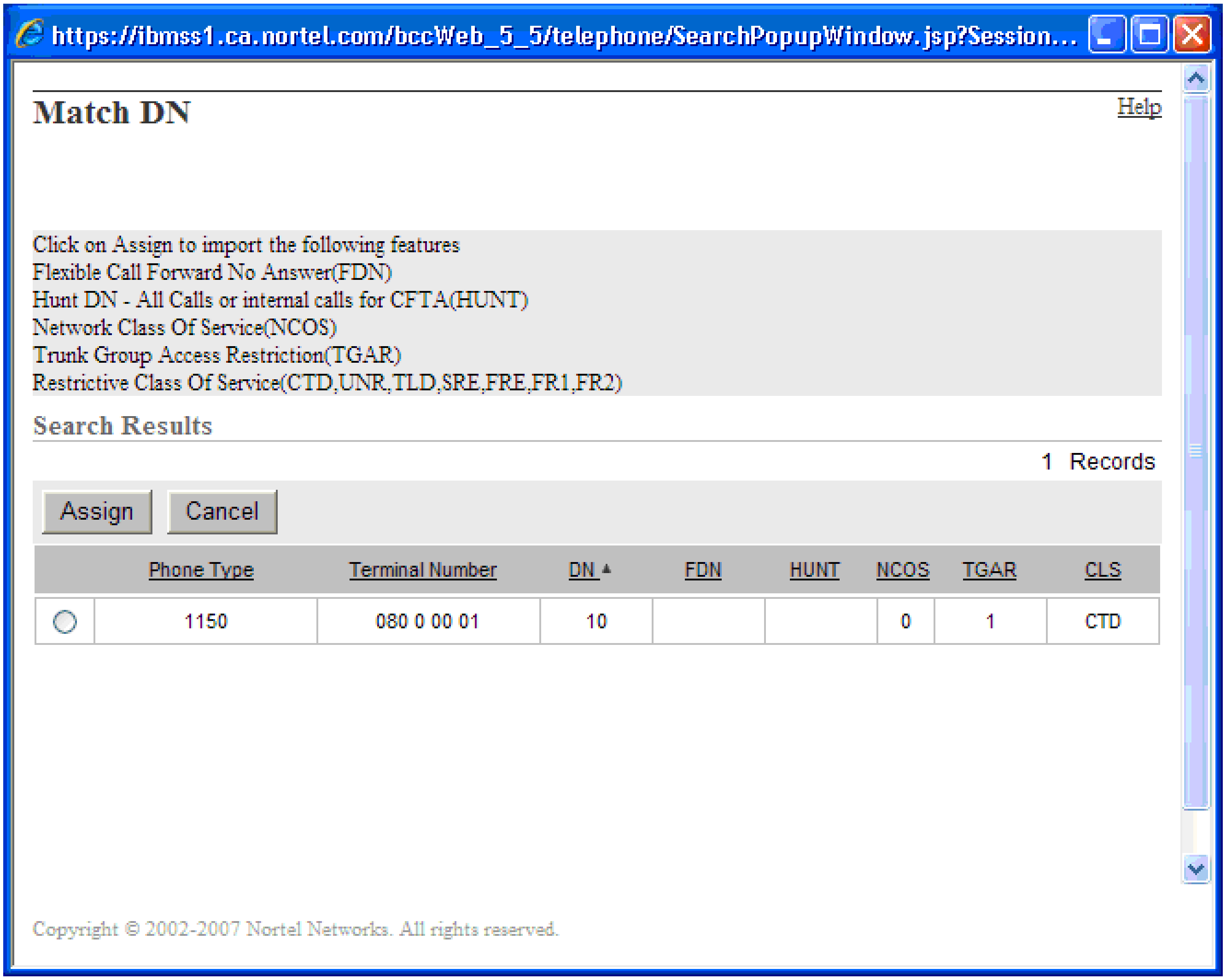


Figure 62: Copy Class of Service from desktop telephone

UEXT-FMCL Configuration

Configuring Features and Keys

1. Logon to ECM using a valid account.
2. Click on the appropriate call server element. Element Manager is launched.
3. Click Phones link from the left navigation pane. The Search for Phones page appears.

Note:

telephone database and call server database must be kept synchronized all the time. Necessary configuration of a customer must already be available for a telephone set to be configured (superloop, customer etc).

4. From the Search for Phones page under the Phones heading, click the Add button. The New Phones page opens.
5. Select UEXT-FMCL-Universal Extension FMCL from the telephone Type drop down menu.
6. Enter the other applicable details under the Type and Options fields and click the Preview button. The telephone Details page appears. The top of the telephone Details page should show the telephone Type as Universal Extension FMCL.

   Note:

   Two features are added under the Features section:

   - UXTY (Universal Extension Type) for which the value is MOBX (Mobile X user). This field cannot be edited.
   - The prompt UXID (Universal Extension ID) is mandatory for Mobile X configuration. It can be up to 16 digits.
7. Navigate to the Keys section of the telephone Details page. Key 0 is configured with the SCR DN. Key 1 is configured as a HOT_P key containing the mobile DN.
8. Enter the required details for Key 0 and Key 1 and click on the Validate button to perform validation of general properties, features and keys.
9. Click the Finish button.

Mobile Directory Service Number

The Mobile Directory Service Number Web page allows users to view, add or delete Mobile Service Directory Numbers. Click Mobile Directory Service Number to open this Web page.

To delete a Mobile Service Directory Number, check one or more of the check boxes and click Delete. The selected entries are deleted and the page is reloaded.

To add a Mobile Service Directory Number, click Add. The Add Mobile Service Directory Number Web page opens.

Enter the appropriate information and click Save.

Make the necessary changes and click Save.

Feature operation

Handoff from Mobile telephone to Desktop telephone

The following diagram depicts a Handoff from a Mobile telephone to a Desktop telephone.

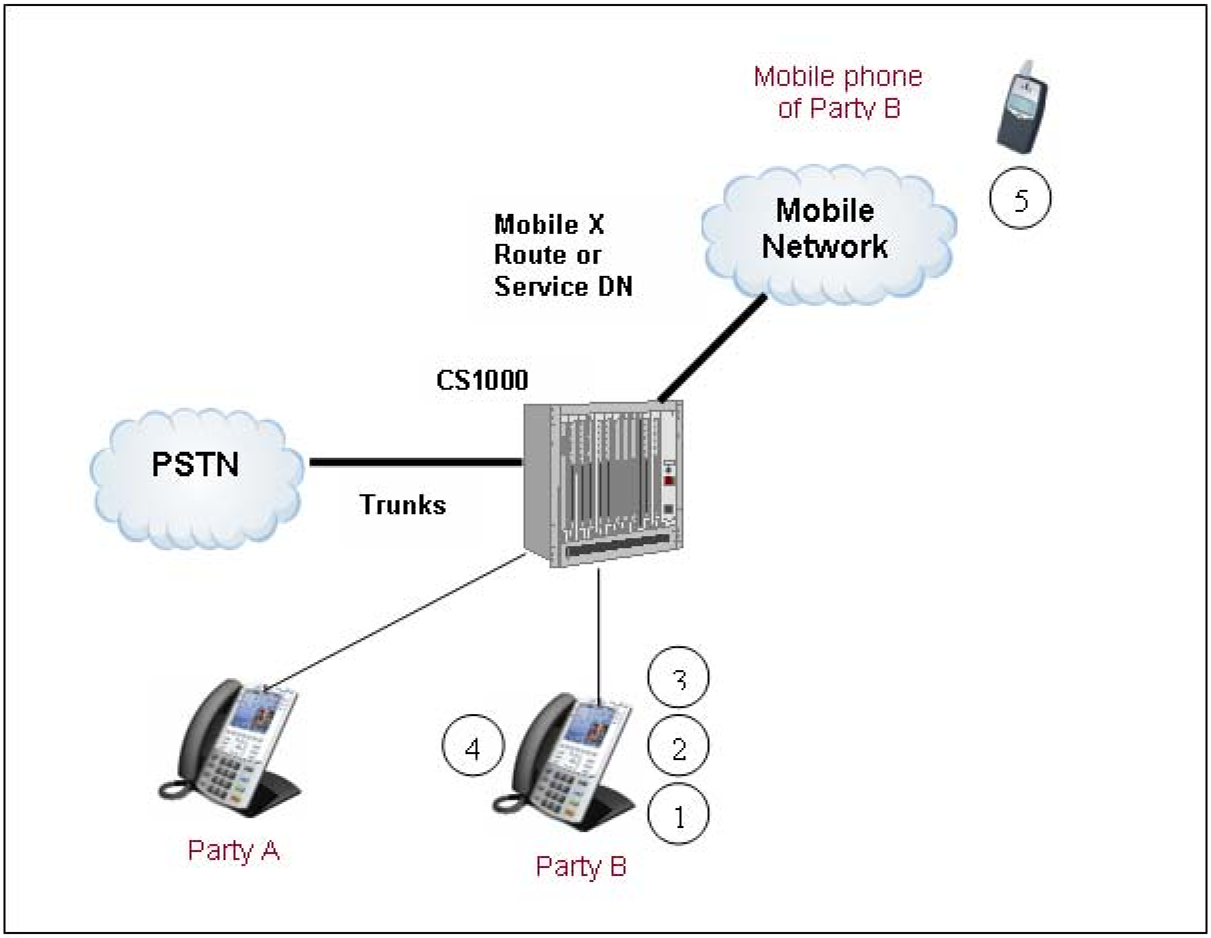


Figure 63: Handoff from Mobile telephone to Desktop telephone

The operation sequence for the Mobile X user is:

1. The user presses the Handoff key on the desktop telephone.
2. The user dials the Station Control Password of the mobile user's UEXT (if any).

3. A conference call is established between Party A, Party B and the mobile telephone of Party B.
4. The user disconnects the call on the mobile telephone.

Handoff from Desktop telephone to Mobile telephone

The following diagram depicts a Handoff from a Desktop telephone to a Mobile telephone.

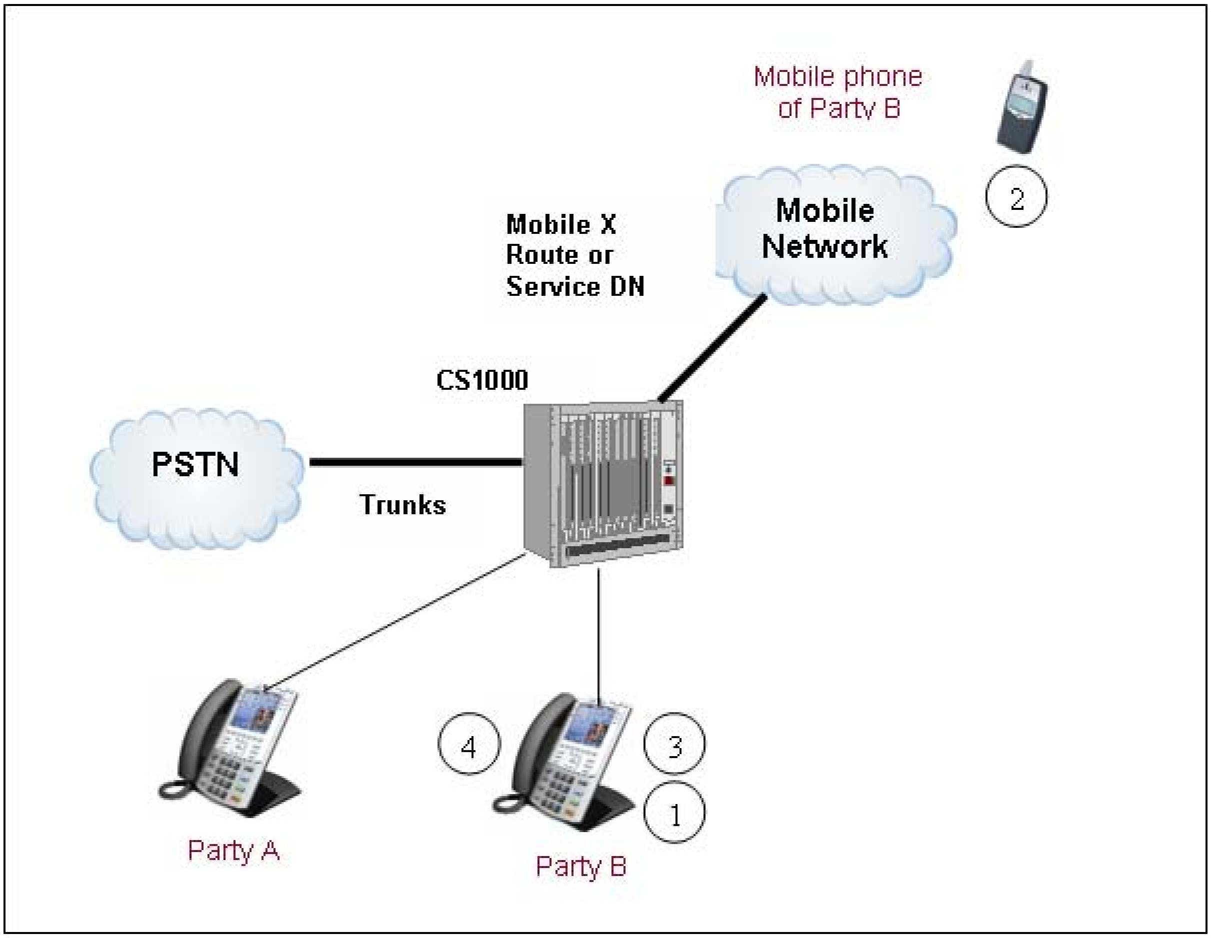


Figure 64: Handoff from Desktop telephone to Mobile telephone

The operation sequence for the Mobile X user is:

1. The user presses the Handoff key on the desktop telephone. The Handoff key lamp is lit and flashing.
2. The user answers the ringing mobile telephone.

3. The Handoff key lamp is off. A conference call is established between Party A, Party B and the mobile telephone of Party B.
4. The user disconnects the call on the desktop telephone.

Call Transfer with consultation

The following diagram depicts a Call Transfer with consultation.

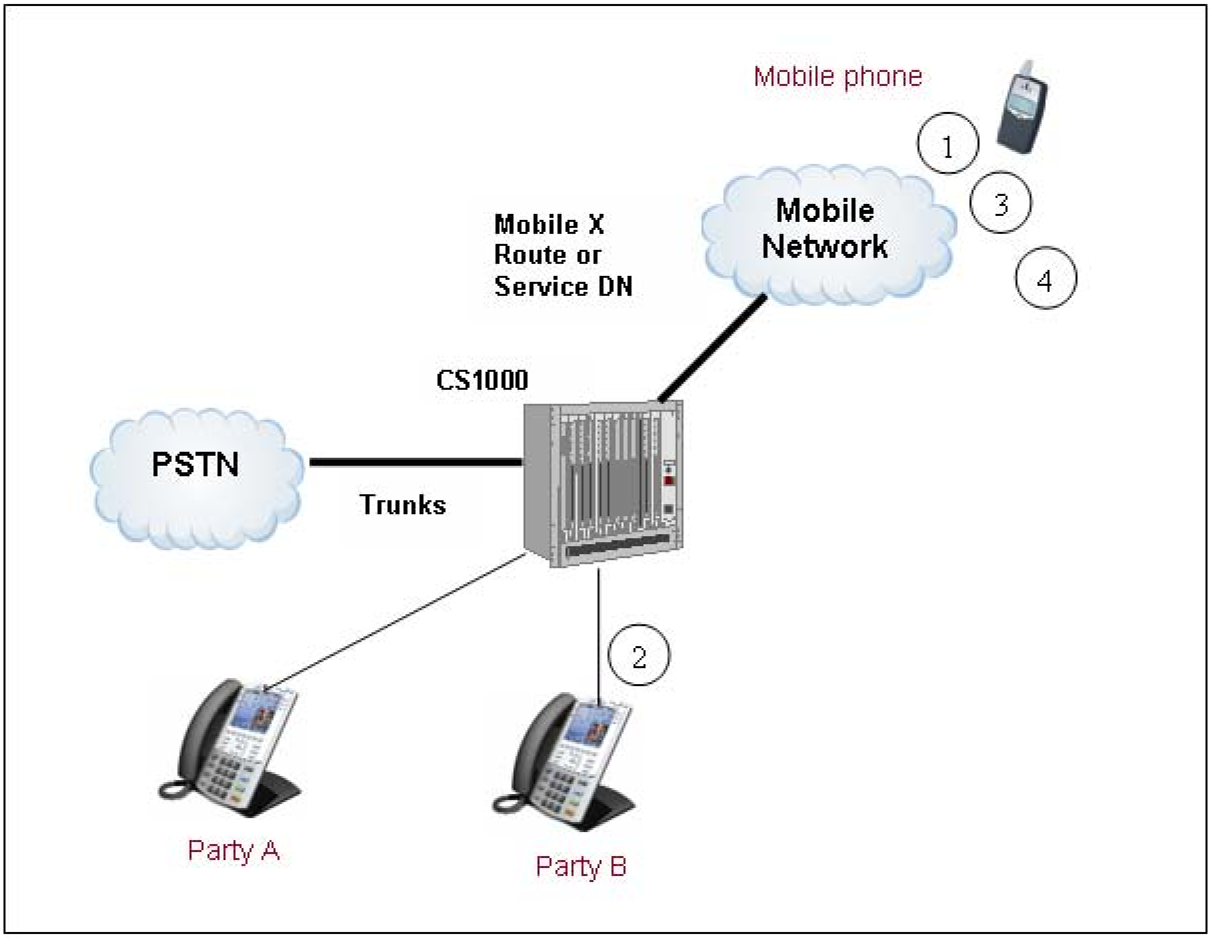


Figure 65: Call Transfer with consultation

The operation sequence is: (assuming the DNB is not busy)

1. While in conversation with another party, the mobile telephone user decides to activate a Call Transfer feature. The mobile telephone user presses MFAC and dials the FFC MTRN to activate the Call Transfer feature, followed by the number where the user wants to transfer the call. This will ring the third party (Party B).
2. Once the call is answered by the third party (Party B), the user can have a consultation with the third party.

3. The mobile telephone user presses the MFAC and dials the MTGL Feature-toggle FFC to toggle between conversations with party A and party B. This step can be omitted.
4. The mobile telephone user presses the MFAC again and enters in the MCOM FFC to complete the transfer.

Blind Transfer

The following diagram depicts a Blind Transfer.

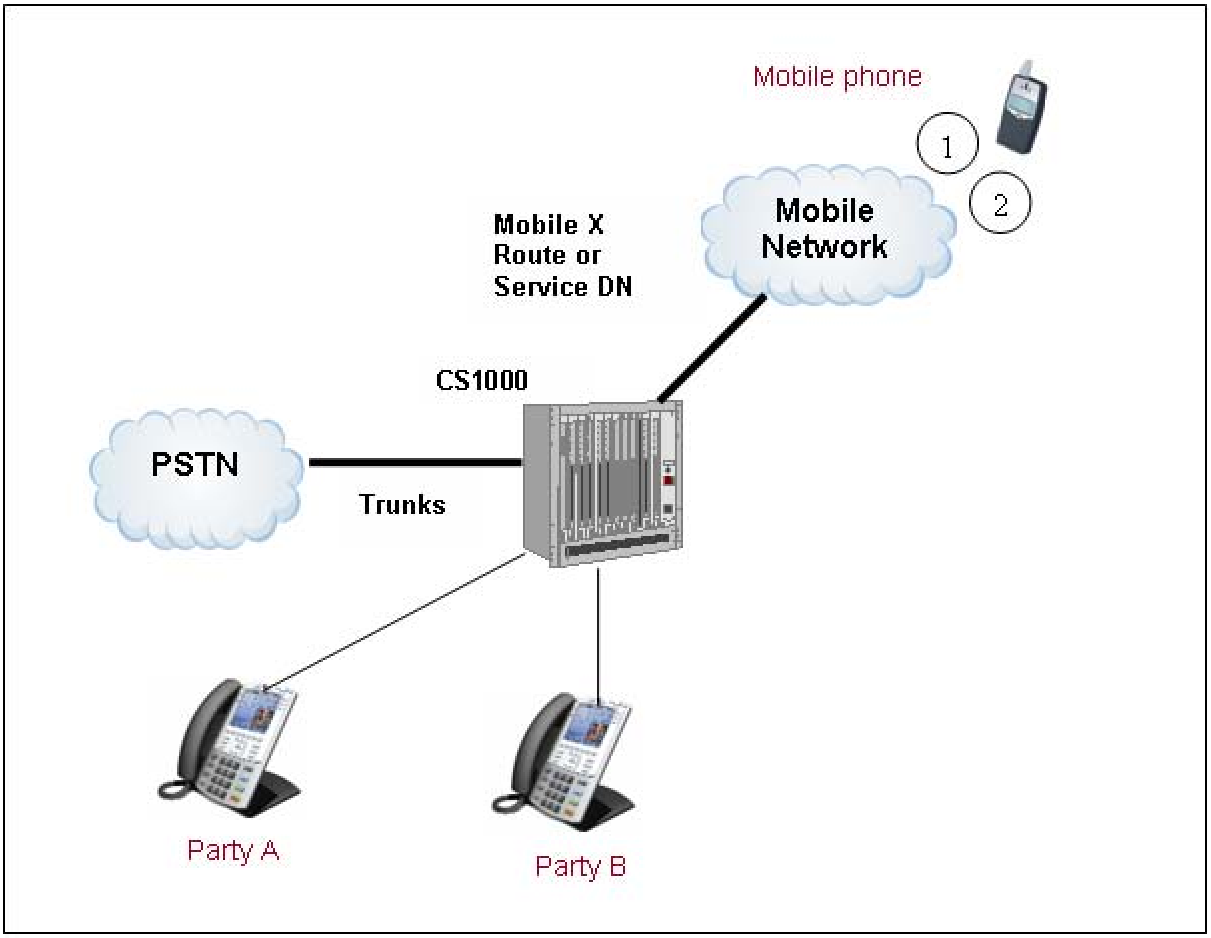


Figure 66: Blind Transfer

The operation sequence is: (assuming the DNB is not busy)

1. While in conversation with another party, the mobile telephone user decides to activate a Call Transfer feature. The mobile telephone user presses the MFAC and

dials the FFC MTRN to activate the Call Transfer feature, followed by the number where the user wants to transfer the call. This will ring the third party (Party B).

2. The mobile telephone user presses the MFAC again and enters the MCOM FFC to complete the feature.

Conference with consultation

The following diagram illustrates a Conference with consultation.

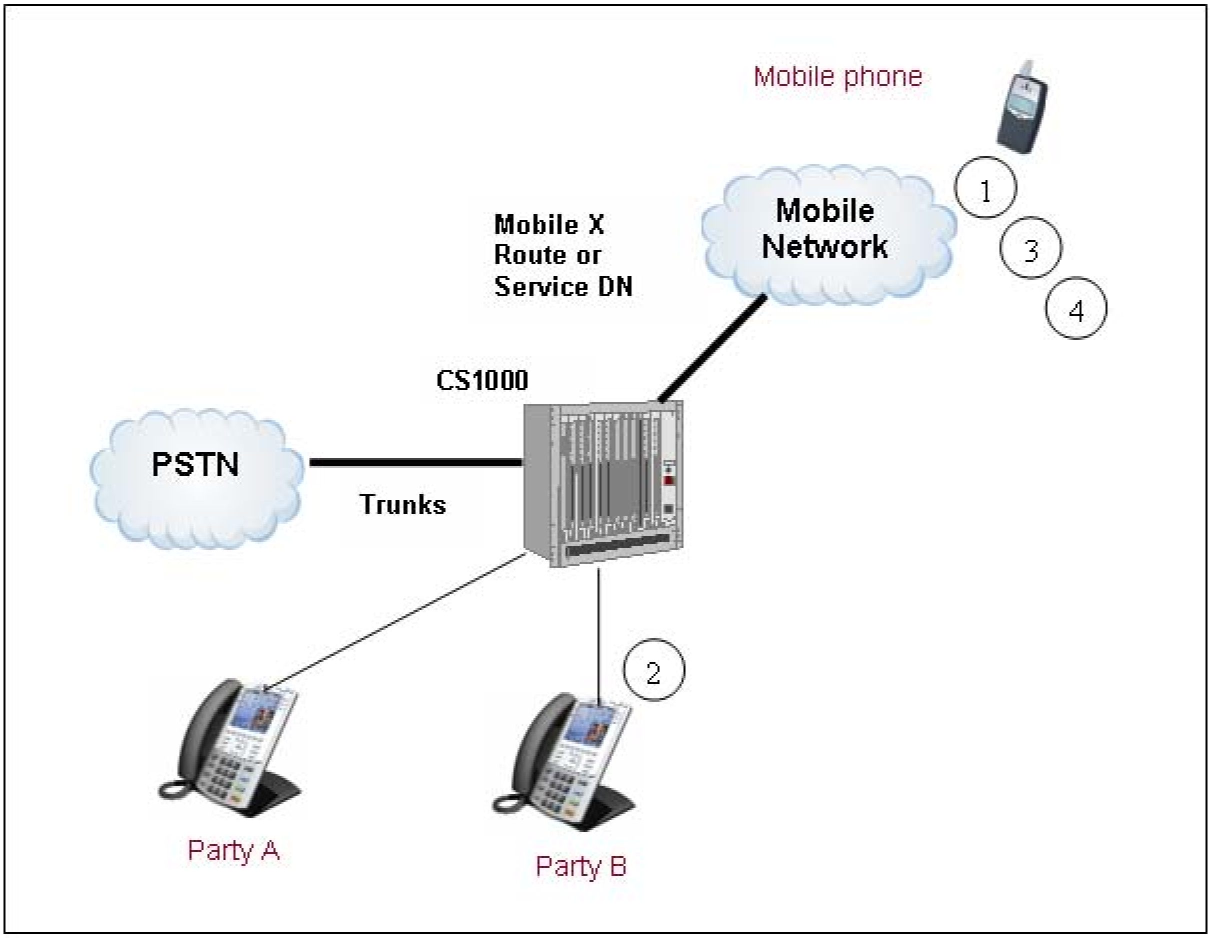


Figure 67: Conference with consultation

The operation sequence is: (assuming the DNB is not busy)

1. While in conversation with another party, the mobile telephone user decides to activate a Conference feature. The user presses MFAC and dials the Mobile Conference Activation (MCFA) FFC to specify the Conference feature to activate

followed by the number which the mobile telephone user wants to add to the conversation. This will ring the third party (party B).

2. Once the call is answered by the third party (party B), the user can have a consultation with the new party.
3. The Mobile X User can press the MFAC and dial the Feature-toggle FFC MTGL to toggle between conversations with party A and party B. This step can be omitted.
4. The Mobile X User can press the MFAC again and enter in the MCOM FFC to complete the feature. A three party conference is established.

Call in Progress feature Activation using MFAC

The following diagram illustrates the sequence of events that take place when a Mobile X user initiates the Call in Progress feature from a mobile telephone using the MFAC.

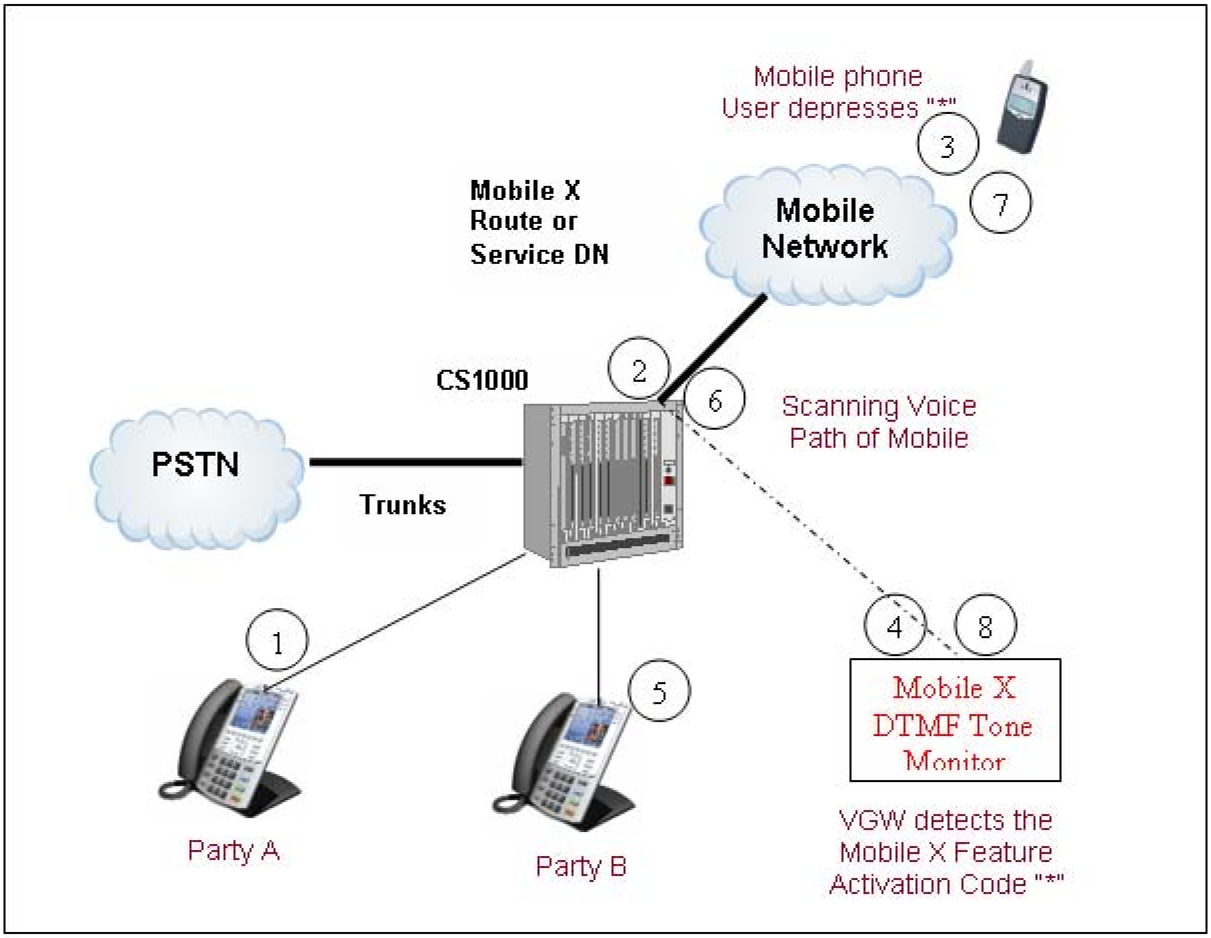


Figure 68: Feature activation example

To activate or cancel the Call in Progress feature, the mobile user must press the MFAC followed by the FFC code and DN. FFC package 139 is used for programming the Call in Progress FFC codes.

The Basic Operation sequence is:

1. A call is setup between a mobile telephone and Party A, another party in a CS 1000 system. The active trunk connected to the mobile telephone is marked as a Mobile X trunk call.
2. The voice path from the Mobile user is monitored to detect a Mobile Feature Activation Code (MFAC).
3. The mobile telephone user activates a Call in Progress feature by pressing MFAC to signal the CS 1000 system.
4. Once the MFAC is detected by the DTMF Tone Monitor in the CS 1000 system, the device sends a signal to notify the CS 1000 system to initiate Call in Progress features. This operation is very similar to the switchhook flash on an analog (500/2500 type) telephone. The active call is placed on hold. A special dial tone is provided.

5. The Mobile X user dials the FFC to specify the feature to activate, for example Call Transfer, and then the number to which the user wants to transfer the call to Party B. This will ring Party B.
6. Once the call is answered, the user can have a consultation with the third party (Party B). This call is monitored for MFAC detection.
7. The Mobile X User presses MFAC again to signal the CS 1000 system for further actions.
8. The DTMF Tone Monitor detects the MFAC and sends a signal to notify the CS 1000 system that a Mobile X User requests further actions. The Mobile User can enter in the FFC to complete the feature activation.

Call Pilot

After you connect to Call Pilot, operation is identical to your desktop phone.

Chapter 78: Multi-language Messaging

Contents

This section contains information on the following topics:

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 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 is displayed in English, because 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 Systeminstallation tools screens
- LD 135 and LD 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

Table 219: 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	(NONE) HELP BOTH	Selects which messages are going to be translated. NONE = Help and Large Systemspecific system messages are printed in English. HELP = Help is printed in the translated language and Large Systemspecific system messages are printed in English. BOTH = Help and Large Systemspecific system messages are printed in the translated language. The translated language printed is dependent on the software packaging.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 79: Multi-Party Operations

Contents

This section contains information on the following topics:

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 telephone, 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:

1. 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.
2. Dial the Toggle Control Digit (TGLD) to exchange the active and held calls.
3. 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 telephones 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, because transfer in the ringing state is allowed, the user may still misoperate 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) telephone 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 telephone (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 telephone 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 is 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 is 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 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 telephones to be programmed for conference, toggle between telephones, and disconnect. However, the toggle function is disabled if a call is transferred to the attendant because of the Attendant Recall with Splitting feature.

Call Forward All Calls

A telephone 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 telephone 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 is 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) telephones 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 telephones 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 is 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 telephone tries to re-ring the transferring telephone, if the transferring telephone 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 which are involved in the Consultation connection.

Camp-On, Forced Override

With Multi-Party Operations (MPO), when a consultation call is made on a telephone equipped with Priority Override, a control digit has to be dialed from the telephone 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 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 telephones. When setting up a conference call, by conferencing one telephone at a time, the display on the conferee telephone is in accordance with the individual telephone Class of Service. If one telephone leaves a three party conference, display information on the remaining telephones is based on the individual Class of Service of each telephone.

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 telephone assigned TSA Class of Service cannot also be assigned ASCA Class of Service, and vice versa; a telephone 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 is 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 is 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. Table 220: LD 15 on page 660

   Configure the Multi-Party Operations parameters in the Customer data block.
2. Table 221: LD 10 on page 663

   Assign Three party Service (TSA) Class of Service to telephones.
3. Table 222: LD 56 on page 663

   Configure Control Dial Tone and recall Tones and Cadences for analog (500/2500-type) telephones and Meridian 1 proprietary telephones.

Table 220: LD 15

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.

Prompt	Response	Description
	(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 telephone 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 telephone 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 telephone 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 telephone 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 If entered in response to AOCS in LD 15, responses are 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.

Prompt	Response	Description
- - 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. 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.

Prompt	Response	Description
		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.

Table 221: LD 10

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 telephone 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 telephone 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.)
...		

Use Table 222: LD 56 on page 663to

Table 222: LD 56

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.

Prompt	Response	Description
...		
- 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 telephones. Default is value assigned to NCAD.
- PBCS		Recall of misoperation ringing tone and cadence for Meridian 1 proprietary telephones.
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.

For more information about administration of tones and cadences, see *Software Input Output - Administration, NN43001-611*.

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) telephone – For the purpose of this document, this term is used to see standard analog (500/2500-type) telephones.

Meridian 1 proprietary telephone – For the purpose of this document, this term is used to see standard SL-1 telephones and to digital telephones (M2000 series and M3900 series).

Bridged telephones – The "Bridging" feature allows the same DN to appear on more than one single line telephone. Bridged telephones share the same TN. Up to eight of these telephones can be bridged, and a maximum of five of this group can be equipped with ringers. An incoming call rings all telephones 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:

  M2317 telephone 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 limitations) on DNx. The Conference key is idled. If the user disconnects during the conference, party A is transferred to party B, subject to normal limitations.

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.

The DNx or secondary DN key can be any Meridian 1 proprietary telephone key capable of holding an independent Directory Number.

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 telephones 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 telephones with DTN Class of Service the mode of dialing control digits depends upon how the telephone has setup the consultation call. If the telephone has used pulse dialing, then the control digits are recognized without recall. If the telephone 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 telephone 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 limitations.
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 limitations.

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 telephone

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 telephones 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 223: 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

Controlling party action	System Response	
	Class of Service	
	XFA	TSA
Dial DISD	Conference	Release Active Party
On-hook	Transfer	Disconnect

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.

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.

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:

1. 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.
2. 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 telephone is rung back.

In the case of1 on page 669 (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 limitation 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 telephone 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 telephones.

With RALL = NO and a 500 telephone (dial impulse) dials a control digit other than DISD, the telephone is given overflow tone indicating that the held party has disconnected. If the 500 telephone 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 telephone is optionally (AAR) rung back immediately. The external party does not receive Ringback Tone while the controlling telephone 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 telephone 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:

1. 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.
2. The user dials the Conference (CNFD) Programmable Control Digit to form a three-party conference. The Consultation connection becomes a Conference connection.
3. 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.
4. 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 2 on page 671. 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 limitations. 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 limitations.

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:

1. STD (Standard) – The operation as it was prior to the introduction of the MPO feature.
2. 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.
3. 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.
4. 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.
5. 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.
6. If the transferring station became busy before recall, the transferred party is disconnected immediately.
7. 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.

8. 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.
9. If the transferring station became busy before recall, the transferred party is routed to attendant immediately.
10. OVF (Overflow) – Overflow Tone is given to the transferred party after the optional number of ring cycles (RCY2).
11. If the transferred call was a Consultation connection, the transferred party is disconnected and the held party is given Overflow Tone.
12. DIS (Disconnect) – The transferred party is disconnected after the optional number of ring cycles (RCY2).
13. If the transferred call was a Consultation connection the transferred party and held party are disconnected.

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:

1. Dialing a vacant number.
2. Dialing a number of a terminal in the maintenance busy or RPE failure state.
3. Access denied.
4. Code Restriction or Toll Restriction.
5. 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 limitations), 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 telephone
    - Network blocking
    - Invalid, restricted and blocked Network Automatic Route Selection (NARS)/Basic Automatic Route Selection (BARS) calls
    - Partial dialing
    - Trunk-to-trunk connection limitations
    - 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 telephone is rung back after Call Transfer misoperation, Recall of misoperation ringing cadence is optionally given to this telephone. A Recall of misoperation cadence for analog (500/2500-type) telephones and Meridian Modular telephones (PCAD) is optionally selectable in LD 56.

Chapter 80: Multi-Party Operations Enhancements

Contents

This section contains information on the following topics:

Feature description

The following enhancements pertain to the Three-party Service capability of Multi-Party Operations (MPO). See 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 telephone 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

Table 224: 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
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.

For more information about the administration of tones and cadences, see *Software Input Output - Administration, NN43001-611*.

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 telephone then receives Overflow Tone. If no digit is dialed, the controlling telephone 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 telephone immediately if the appropriate option (either AAR or DAR) is active. The external party does not receive ringback tone while the controlling telephone is being rung.

This enhancement allows a ringback tone to be provided to the external party when the controlling telephone is being rerung.

Chapter 81: Multiple Appearance Directory Number Redirection Prime

Contents

This section contains information on the following topics:

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, see 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

- 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 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

  - 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

  - 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

  - 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

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 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 instructions (not the MARP TN), as shown in Figure 69: CFW and MARP on page 687.

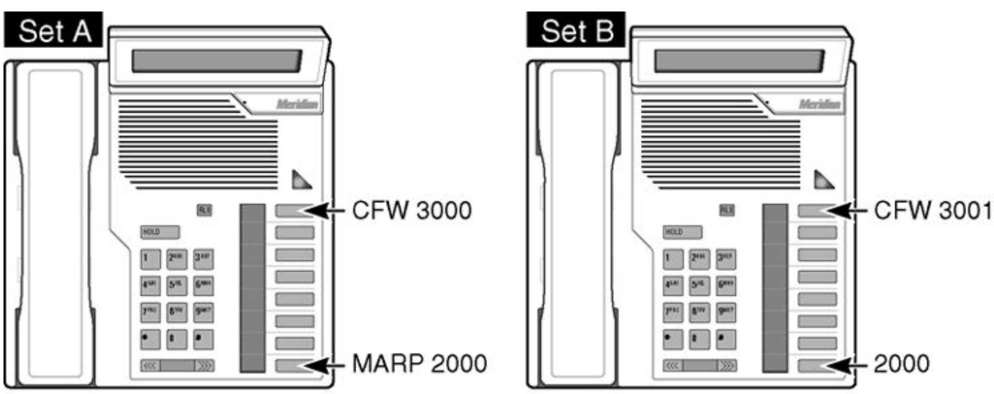


Figure 69: CFW and MARP

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. See Figure 70: MARP control on page 688.

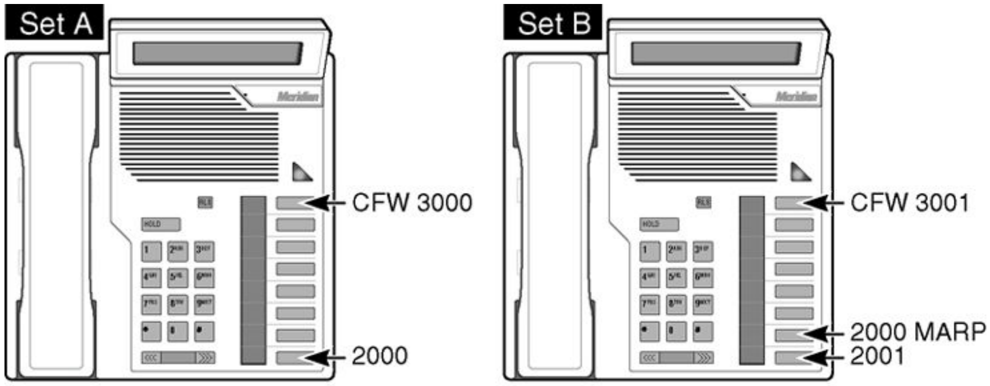


Figure 70: MARP control

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 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 SCUPPER, 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 telephone is allocated the same prime DN, the Hot Type I call will terminate on the telephone designated as the Multiple Appearance Redirection Prime (MARP). If the MARP DN is not the prime DN on the telephone, or if the telephone designated as the MARP DN is not a Meridian 1 proprietary telephone, the first Meridian 1 proprietary telephone with the prime DN is used. If none of these conditions are met, the call will terminate as a non-Hot Line call and the calling party is 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 DNs 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. Table 225: LD 17 on page 692

   Activate or deactivate MARP.

2. Table 226: LD 10 on page 692

   Add an analog (500/2500-type) telephone with a Single Appearance DN.

3. Table 227: LD 10 on page 692

   Add an analog (500/2500-type) telephone with a Multiple Appearance DN.

4. Table 228: LD 10 on page 693

   Change an analog (500/2500-type) telephone with a Multiple Appearance DN.

5. Table 229: LD 11 on page 693

   - Add a telephone with a Single Appearance DN.

6. Table 230: LD 11 on page 694

   Add a telephone with a Multiple Appearance DN.

7. Table 231: LD 11 on page 694

   Change a telephone with a Multiple Appearance DN.

8. Table 232: LD 10 on page 695

   Remove a MARP TN.

9. Table 233: LD 11 on page 695

   Remove a MARP TN.

10. Table 234: LD 20 or LD 22 on page 695

    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.

Table 225: LD 17

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.

Table 226: LD 10

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	500	500/2500 telephone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
DN	xxx...x	Directory Number.
- MARP		MARP prints on the next line indicating this TN is the MARP for DN xxxx.

Table 227: LD 10

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	500	500/2500 telephone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.

Table 228: LD 10

Prompt	Response	Description
REQ:	CHG	Modify existing data.
TYPE:	500	500/2500 telephone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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).
- MARP	(NO) YES	(Do not) set the MARP to this TN.

Table 229: LD 11

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Prompt	Response	Description
- MARP		MARP prints on the next line indicating this TN is the MARP for DN yyyy.
KEY		Reprompts until <CR> is entered.

Table 230: LD 11

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.
KEY		Reprompts until <CR> is entered.

Table 231: LD 11

Prompt	Response	Description
REQ:	CHG	Modify existing data
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Prompt	Response	Description
- 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) is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to the working TN.
KEY		Reprompts until <CR> is entered.

Table 232: LD 10

Prompt	Response	Description
REQ:	OUT	Remove data from the system.
TYPE:	a...a	Telephone type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.

Table 233: LD 11

Prompt	Response	Description
REQ:	OUT	Remove data from the system.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.

Table 234: LD 20 or LD 22

Prompt	Response	Description
REQ	PRT	Print information.
TYPE	TNB (DNB)	Terminal Number data block. (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 2016			
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 2016

  KEY 00 MCR 2000 MARP

  01 MRK

Feature operation

No specific operating procedures are required to use this feature.

Chapter 82: Multiple Appearance Directory Number

Contents

This section contains information on the following topics:

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 telephones 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 first primary appearance terminal number.

Call Detail Recording on Redirected Incoming Calls

If the DN of the telephone forwarding the call is a Multiple Appearance DN, the Terminal Number of the telephone 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 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 is received by the user.

Calling Party Name Display Denied

For a ringing call to a Multiple Appearance DN, the name on the calling telephone display can be suppressed by configuring any of the Terminal Numbers with NAMD Class of Service. The digit display on the calling telephone cannot be suppressed – the called digits are displayed even though the Class of Service on any of the Terminal Numbers is DIGD. The called telephone display is subject to the Class of Service of the calling party. For an established call to a Multiple Appearance DN, the calling telephone display is subject to the Class of Service configured for the answering telephone. The answering telephone display only is subject to the Class of Service of the calling party – the displays of the other telephones 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 telephone, 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)

Because the ANI category is defined on a per telephone 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 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 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 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.

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 telephone must be assigned to the hunt group. This supervisor telephone 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 telephone controls the status of the MADN and thus the group hunt treatment. If the supervisor telephone 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 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 telephones can be defined as an SCR (single call ringing) key or as an MCR (multiple call ringing) key. For those DNs (keys on MADN telephones) 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 telephones 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

Because 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 telephones 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 telephone on which it is active).

Three Wire Analog Trunk - Commonwealth of Independent States (CIS)

Because the ANI category is defined on a per telephone 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

Table 235: LD 11 - Assign a Multiple Appearance Directory Number key.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u c u	Terminal Number For Large Systems For Small Systems
KEY	xx MCN yyy...y	Add a multiple-call non-ringing DN key, where:
	xx MCR yyy...y	xx = key number, and yyy...y = DN. 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.

Chapter 83: Multiple Console Operation

Contents

This section contains information on the following topics:

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. Table 236: LD 12 on page 712

   Select attendant console number.
2. Table 237: LD 15 on page 712

   Select Supervisory Console.
3. Table 238: LD 93 on page 712

   Configure Multi-Tenant Service.

Table 236: LD 12

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
...		
ANUM	1-63	Attendant Number.

Table 237: LD 15

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	ATT_DATA	Attendant console options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
...		
SPVC	(0)-63	Supervisory Console.

Table 238: LD 93

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.

Chapter 84: Multiple Customer Operation

Contents

This section contains information on the following topics:

Feature description

The system can serve up to 32 (customer numbers 0 to 31) individual customers from the same machine. The system software supports 100 customer groups (numbered 0 to 99). Customers have their own features, limitations, 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.

Chapter 85: Multi-Site Mobility Networking

Contents

This section contains information on the following topics:

Feature description

Multi-Site Mobility Networking (MSMN) allows a Nortel DECT Handset 4060 user to place and receive calls at any MCDN node. When the handset user visits a MCDN node, the MSMN feature automatically does the following:

- detects the visiting handset when the handset is on
- forwards calls to the visiting handset from the user home node

The call forward dial tone gives an indication when MSMN activation is not successful. The user can turn the handset off and on again to reactivate the MSMN feature.

The MSMN feature requires concentrated DMCs. The DMCs must be 8D to support concentration. In a non-concentrated system, each handset is configured to a DMC TN. A non-concentrated DMC has 32 handset TNs assigned to 32 time slots, and is non-blocking. In a concentrated system, each handset is configured to a Phantom TN on a Phantom loop. Concentration allows up to 510 handsets to share the DMC 32 time slots, (a potentially blocking system).

Separate Nortel Integrated DECT (DECT) systems on a Meridian circuit-switched network can be either concentrated or non-concentrated.

Note:

DECT in this chapter refers to DMC DECT, not SIP DECT.

Operating parameters

The MSMN feature cannot support a mix of concentrated DMCs and non-concentrated DMCs within the same DECT system. All DMCs must have at least one handset configured.

When users visit a Companion DECT system, other than their home system, they lift the handset, wait to hear a dial tone, and hang up. This forces the registration process to finish successfully; the state of the telephone is fully synchronized with the state of the visited system.

Feature interactions

CallForward All Calls from a MADN handset

A MADN handset at a remote node can activate Call Forward All Calls at the home node. When the handset shares a DN with another telephone, the lamp associated with the CFW key on the shared DN telephone lights up. If the handset is not the MARP, the shared DN MARP telephone user can cancel Call Forward All Calls. If the handset is the MARP, the handset user can override any call forwarding that is setup from other shared DN telephones.

Card audit

Card audit does not work with Phantom TNs.

Network Message Service

The MSMN feature does not change the handling of unanswered network calls. The Meridian Mail or CallPilot 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. Table 239: LD 10 on page 720

   Add a new DCS handset.
2. Table 240: LD 10 on page 721

   Copy DCS handsets on DMC.
3. Table 241: LD 10 on page 721

   Remove DCS handsets.
4. Table 242: LD 10 on page 722

   Convert handset type 500 to DCS.
5. Table 243: LD 20 on page 722

   Print actual DMC TN and Phantom DMC TN list.
6. Table 244: LD 81 on page 723

   Print DCS features.

The tasks 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 handsets on the DMC.
5. Presubscribe the visiting handset one time at the MCDN node.

   Subscription includes both overlay configuration and DECT Manager configuration. For DECT Manager configuration, see *DECT Fundamentals, NN43120-114*.

Table 239: LD 10

Prompt	Response	Description
REQ:		
	NEW	Add a Digital Cordless Set.
	NEW 1-255	The generation of new DCS units stops when the maximum Index of 509 is reached on a single DMC, or Phantom TNs on the system run out, or the WRLS License limit is reached. All new DCSs 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 DECT Handsets 4060.
		If TYPE = DCS, the system allocates the next available VTN, 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		Terminal number.
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Prompt	Response	Description
MWUN	32	Maximum number of Wireless UNits. Entry defaults to 32 with no user input. Value cannot be CHG'ed. If MWUN = 32, CDEN automatically changes to 8D, and prints as an 8D unit.
DMC		DECT Mobility Controller Location.
	l s c	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card.
	c	Format for Small System and Media Gateway 1000B where c = card.
INDX	0 .. 509	DMC index to map the Phantom 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.

Table 240: LD 10

Prompt	Response	Description
REQ:	CPY 1-32	CPY n = The generation of new units stops when the maximum index of 509 is reached on a single DMC, Phantom TNs on the system run out, or the WRLS License limit is reached. All DCSs must be on the same DMC.
DMC		DECT Mobility Controller Location.
	l s c	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card.
	c	Format for Small System and Media Gateway 1000B where c = card.

Table 241: LD 10

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		DECT Mobility Controller Location.
	l s c	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card.

Prompt	Response	Description
	c	Format for Small System and Media Gateway 1000B where c = card.

Table 242: LD 10

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.

To convert from concentrated to non-concentrated, OUT all DCS units and resubscribe the handsets.

The CDCS command can also be used on MCMO type handsets.

The CDCS conversion routine prints each TN as the TN is moved, in the following format:

500 TN l s c 00 = DCS TN L' S' C' Index#.

- L' S' C' = Phantom TN
- Index# = default of the unit number of the 500 type telephone

Table 243: LD 20

Prompt	Response	Description
REQ	PRT	Request.
TYPE	DCS	Digital Cordless Set.
TN		Terminal Number for actual DMC.
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
		Phantom Terminal Number.
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.

The print routine outputs the following format:

```
INDX Index # VTN lll s cc uu
```

• Index # = Index number of Phantom TN

• lll s cc uu = Phantom TN of unit

Table 244: LD 81

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
```

• 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 on the handset within the coverage range of a visited Companion DECT system.
2. Enter the coverage range of a visited DECT system from another DECT system with the handset on.
3. Wait for the telephone to be registered within the coverage range of the visited Companion DECT system. Lift the handset, wait to hear a dial tone, and hang up to ensure that the registration process finishes successfully.

To deactivate the MSMN feature:

1. Turn off the handset within the coverage range of the visited Companion DECT system. (The handset must have the DECT Detach feature.)
2. Turn on the handset 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.)

Chapter 86: Multi-Tenant Service

Contents

This section contains information on the following topics:

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 71: Tenant-to-Tenant access on page 727, 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 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.

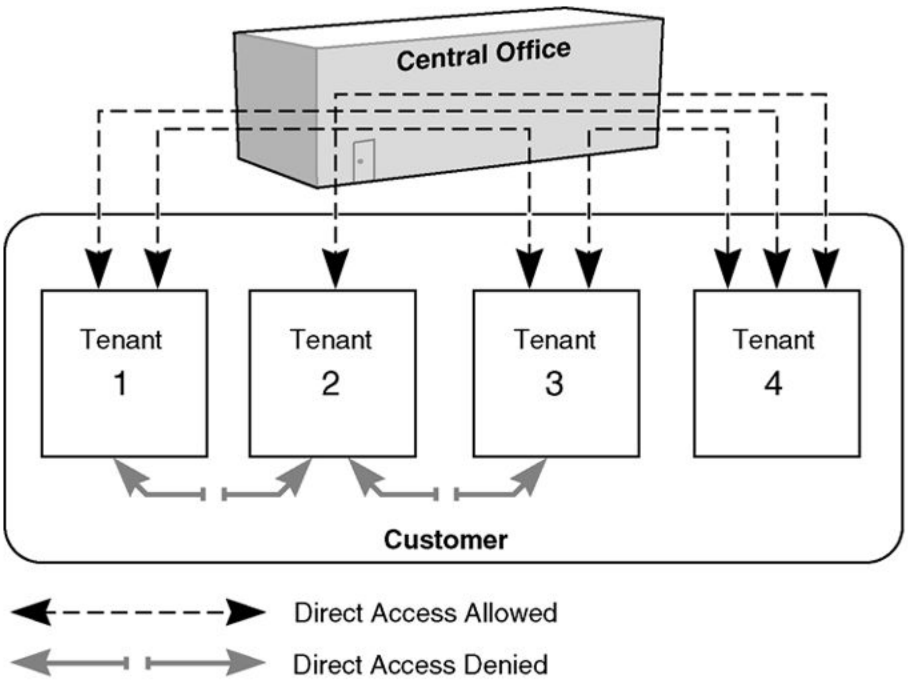


Figure 71: Tenant-to-Tenant access

As shown in Table 245: Tenant-to -Tenant Access allowed or denied on page 727, Tenant-to-Tenant Access allows or denies tenants of the customer:

Table 245: 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 outgoing routes.

A tenant can have private outgoing trunk routes assigned. This is done by denying all other tenants access to the routes.Figure 72: Tenant-to-Trunk Route Access on page 728 shows a diagram of the following tables.

Table 246: 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 72: Tenant-to-Trunk Route Access on page 728, Tenants 2 and 3 share access to route 2.

Table 247: 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

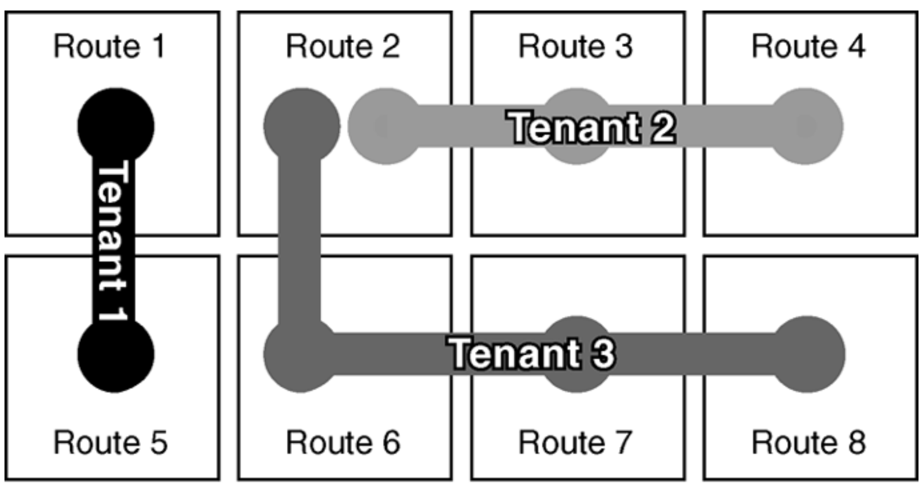


Figure 72: Tenant-to-Trunk Route Access

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.

Tenant-to-Attendant Access (Internal Calls)

Tenant-to-Attendant Access specifies which Attendant Console Group receives automatic presentation of a tenant 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, see Attendant console features on page 729.

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 , 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 248: Typical attendant group arrangement on page 730, 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 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 248: Typical attendant group arrangement on page 730, incoming calls on route 2 are automatically presented to Attendant Console Group 5 (console 7 only).

Table 248: 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	

Attendant group number	Attendant consoles	Incoming Trunk routes	Tenant
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 Music trunks.

Tenants can receive private or shared access to the Paging routes configured for their customer.

All tenants can access their customer 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 telephone 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 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 249: CDR record types and descriptions on page 733.

Table 249: 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 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 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 telephone (A) who picks up a parked call, and party (B), who parked the call, is enforced as follows:

- If B is a telephone, 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, telephone 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 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 Basic Alternate Route Selection, Network Alternate Route Selection (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 Large System.

Network Speed Call

All tenants have access to their customer Network Speed Call (NSC) lists. Any route selected by NSC must have Tenant-to-Route Access allowed.

Off-Hook Queuing

Off-Hook Queuing (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 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 telephone 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 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 IMS.

Integrated Voice Messaging System (IVMS)

Tenants can share or be denied access to their customer 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 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 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 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 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 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. Table 250: LD 93 on page 744

   Enable, disable, or print Multi-Tenant Service for a specified customer.
2. Table 251: LD 93 on page 744

   Allow, deny, or print Tenant-to-Tenant Access for a specified tenant.
3. Table 252: LD 93 on page 744

   Allow, deny, or print Tenant-to-Route Access for a specified trunk route.
4. Table 253: LD 93 on page 745

   Add Console Presentation Group.
5. Table 254: LD 93 on page 745

Assign Tenant-to-Attendant Console access.

6. Table 255: LD 93 on page 746

   Assign Attendant Console group number.

7. Table 256: LD 10 on page 746

   Add Multi-Tenant Service assignments on analog (500/2500) telephones.

8. Table 257: LD 11 on page 746

   Add Multi-Tenant Service assignments on proprietary telephones.

Table 250: LD 93

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.

Table 251: LD 93

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.

Table 252: LD 93

Prompt	Response	Description
REQ	CHG PRT	Change, or print.
TYPE	RACC	Tenant-to-Route Access Data Block.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
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

Table 253: LD 93

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.

Table 254: LD 93

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.

Table 255: LD 93

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		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System Media Gateway 1000B.
AGNO	0-63	Attendant Console group number.

Table 256: LD 10

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
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.

Table 257: LD 11

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CLS	(TEND) TENA	Tenant service (denied) (station shares customer resources and is a non-tenant). Tenant service allowed.

Prompt	Response	Description
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.

Chapter 87: Music

Contents

This section contains information on the following topics:

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 Intelligent Peripheral Equipment (IPE). 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). For further information, see *Automatic Call Distribution Fundamentals, NN43001-551*.

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 757).

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. Table 258: LD 17 on page 753

   Enable conference loops for Music on Hold.
2. Table 259: LD 16 on page 753

   Enable a music route.
3. Table 260: LD 14 on page 754

   Enable a music trunk.
4. Table 261: LD 16 on page 754

   Enable Music on Hold for trunk routes.
5. Table 262: LD 23 on page 755

   Enable Music for an Automatic Call Distribution Directory Number.

Table 258: LD 17

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.

Table 259: LD 16

Prompt	Response	Description
REQ	CHG	Change.

Prompt	Response	Description
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.
All other prompts can be set to default values.		

Table 260: LD 14

Prompt	Response	Description
REQ	NEW	New.
TYPE	MUS	Music trunk.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.
CFLP	0-158	Conference loop assigned to music in LD 17.

Table 261: LD 16

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
TKTP	COT DID FEX TIE WAT	Route type.

Prompt	Response	Description
MUS	(NO) YES	Music on Hold (is not) or is to be provided for this trunk route.
MRT	xxx	Music route number.

Table 262: LD 23

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.

Chapter 88: Music, Enhanced

Contents

This section contains information on the following topics:

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 263: Features vs. no Music, Music, and Enhanced Music on page 757.

Table 263: 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

	Without Music		Music		Enhanced Music	
	Sets	Trunks	Sets	Trunks	Sets	Trunks
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 749.

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.

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. Table 264: LD 17 on page 760

   Add or change Conference loops for Music on Hold.
2. Table 265: LD 15 on page 760

Enable Music Customer Data Block.

3. Table 266: LD 16 on page 760

   Enable a Music route.

4. Table 267: LD 14 on page 761

   Enable a Music trunk. At least one Music trunk per network group is required for each customer requiring Music.

5. Table 268: LD 16 on page 761

   Enable Music on Hold for trunk routes.

Table 264: LD 17

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).

Table 265: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CDB FTR	Customer Data Block Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Small System and Media Gateway 1000B.
- MUS	(NO) YES	Enhanced music for telephones.
- MUSR	0-511	Music route for telephones.

Table 266: LD 16

Prompt	Response	Description
REQ	NEW	New

Prompt	Response	Description
	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.
All other prompts can be set to default values.		

Table 267: LD 14

Prompt	Response	Description
REQ	NEW CHG	New Change
TYPE	MUS	Music trunk.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B.
CFLP	0-158	Conference loop assigned to music in LD 17.

Table 268: LD 16

Prompt	Response	Description
REQ	NEW CHG	New 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.

Chapter 89: Music Broadcast

Contents

This section contains information on the following topics:

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

A License limit is introduced for the Music Broadcast feature. This limits the total number of Music Broadcast connections allowed on a system. The License limit can be allocated over different Music routes and trunks. A License 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 Incremental Software Management information for Music Broadcast connections. The existing SLT command prints the License information for the system.

The License header in LD 14 indicates the number of Music Broadcast connections allowed for the system. The LD 14 header contains the following information:

- The USED parameter represents the maximum number of simultaneously used music connections because the last system load.
- The TOT parameter is the License limit for Music Broadcast connections.
- AVAIL is the difference between the TOT value and the USED value (AVAIL = TOT - USED).

The existing TN information shown in the License 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 License limit at SYSLOAD. The Music Broadcast License limit pertains to Music Broadcast connections only and not to TNs.Figure 73: License header in LD 14 on page 765 is an example of the updated header:

TNS	AVAIL: xxxxx	USED: xxxxx	TOT: xxxxx
MUS CON	**AVAIL: xxxxx**	**USED: xxxxx**	**TOT: xxxxx**

Figure 73: License header in LD 14

Customers can modify License parameters via keycode. A keycode is a machine-generated digitally signed list of customer capabilities and authorized software release. A security keycode scheme protects License parameters.

To expand License 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, see *Communication Server 1000M and Meridian 1 Large System Upgrade Procedures, NN43021-458*.

For more information about Incremental Software Management, see Incremental Software Management.

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 License limit.Figure 74: TFC111 Report for a broadcasting music route on page 766 is an example of the customer report, TFC111, for Music Broadcast routes.

0200 (System ID)	TFC111	
000 (Customer number)		
030 (Route number)	MUS (Trunk type)	
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)

Figure 74: TFC111 Report for a broadcasting music route

Operating parameters

Music Broadcast requires any Music trunk and an external music source or a Nortel Networks Integrated Recorded Announcer card (NTAG36). Integrated Recorded Announcer 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 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 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 License 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 License limit. The License 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.

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. Table 269: LD 16 on page 769

   Change an existing non-broadcasting Music route to a broadcasting Music route.
2. Table 270: LD 16 on page 770

   Enable Conference-based Music route.
3. Table 271: LD 14 on page 770

   Configure Conference-based Music trunks.

Music Broadcast

Table 269: LD 16

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Small System and Media Gateway 1000B.
TKTP	MUS	Music Trunk Data Block.
ICOG	OGT	Outgoing only Trunk.
...		
BDCT	YES	Allow Broadcast capability. NO = Deny Broadcast capability (default), except for CS 1000E where default is YES. If BDCT = YES, no conference loop is required. Each Music trunk has 64 broadcast connections.

Conference-based Music

Table 270: LD 16

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
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.

Table 271: LD 14

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	MUS	Music trunk.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
	c u	Format for Small System and Media Gateway 1000B where c = card and u = unit.
...		
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
	0-127 1-4000	Range for Small System and Media Gateway 1000B
...		
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.

Chapter 90: Music on Hold, Set-based

Contents

This section contains information on the following topics:

Feature description

Set-based Music on Hold provides the option to configure different music for callers on hold based on the type of phone involved in the call. If a call is placed on hold, the destination side listens to Music on Hold as defined in the TN block configuration.

When configuring a phone for Set-based Music on Hold, use the Music Route number (MRT) prompt in LD 11 to configure standard or IP-based music Route Data Blocks. You can apply the same MRT number to multiple phones and configure it at both the Customer Data Block (CDB) and Route Data Block (RDB) levels. Use the Class of Service prompts SBMA (Set-based Music Allowed) and SBMD (Set-based Music Denied) to allow or deny the feature.

This feature supports all possible combinations of phone types. The following list contains examples of supported combinations:

- TDM phone with Standard Digital Music on Hold (Miran card)
- TDM phone with IP-based Music on Hold
- IP phone with Standard Digital Music on Hold (Miran card)
- IP phone with IP-based Music on Hold

Note:

If a phone has an empty MRT and a CLS of SBMA, music is not provided even if the CDB or RDB prompts have music configured. If a phone has a defined MRT and a CLS of SBMD, music is provided as defined in the CDB or RDB.

Operating parameters

There are no specific operating parameters for this feature.

Feature interactions

There are no feature interactions specific to this feature.

Feature packaging

Set-based Music on Hold requires the following packages:

- Music (MUS) package 44
- Recorded Announcement (RAN) package 7

Feature implementation

Assigning a Music Route number to a phone

Table 272: LD 11 — Assigning a MRT to a phone

Prompt	Response	Comment
REQ	aaa	Request (aaa = NEW or CHG)
TYPE	bbb	Type of terminal. It can be any BCS or PBX phone.

Prompt	Response	Comment
CUST	xx	
...		
MRT	N	N - Music route number. Route type is to be MUS for standard phones or IMUS for IP Phones. By default, MRT is empty.
ERL	yyy	
...		

Enabling Set-based Music on Hold for a phone

Table 273: LD 11 — Enabling set-based Music on Hold

Prompt	Response	Comment
REQ	aaa	Request (aaa = NEW or CHG)
TYPE	bbb	Type of terminal. It can be any BCS or PBX phone.
CUST	xx	
CAC_MFC	...	
CLS	(SBMD)/SBMA	Set Based Music Denied/ Allowed
CPND_LANG	...	
...		

Feature operation

No specific operating procedures are required to use this feature.