



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

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Chapter 1: New in this release

The following sections detail what is new in the *Features and Services Fundamentals – Book 4 of 6, NN43001–106* for Avaya Communication Server 1000 Release 7.5.

- [Features](#) on page 31
- [Other changes](#) on page 31

Features

The following features have been updated for Avaya Communication Server 1000 Release 7.5:

- [IP Ad Hoc Conference](#) on page 197
- [IP Attendant Console](#) on page 203
- [IP Music](#) on page 215
- [IP Recorded Announcements](#) on page 255
- [IP Tone](#) on page 263

Other changes

The following have been renamed and moved from this publication to *Avaya Features and Services Fundamentals - Book 1 of 6, NN43001-106*.

Old name	New name
IP Phone 1210 Last Number Redial SoftKey	Avaya 1210 IP Deskphone Last Number Redial soft key
M3900 Full Icon Support	Avaya 3900 Series Digital Deskphones Full Icon Support
M3900 Set-To-Set Messaging	Avaya 3900 Series Digital Deskphones Set-to-Set Messaging
M3900 Single Site Virtual Office	Avaya 3900 Series Digital Deskphones (Single Site) Virtual Office
Nortel Integrated DECT	Avaya Integrated DECT

Revision History

April 2012	Standard 05.10. This document is up-issued to include additional information on the Mobile Extension Handoff feature.
December 2011	Standard 05.09. This document is up-issued to support the following: • Removal of End of Life (EoL) and Manufactured Discontinued (MD) hardware content and associated diagrams. • Changes made to the Multiple Appearance Directory Number chapter.
October 2011	Standard 05.08. This document is up-issued to support Communication Server 1000 Release 7.5. Branch Office/ Survivable Remote Gateway (BO/SMG) operation is added in Trunk to Trunk Connection under Multi-Party Operation (MPO) feature interactions.
August 2011	Standard 05.07. This document is up-issued to support Communication Server 1000 Release 7.5. Changes made to the Multiple Appearance Directory Number chapter.
May 2011	Standard 05.06. This document is up-issued to support Communication Server 1000 Release 7.5.
May 2011	Standard 05.05. This document is up-issued to support Communication Server 1000 Release 7.5.
February 2011	Standard 05.04. This document is up-issued to remove legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.
December 2010	Standard 05.03. This document is up-issued to update the Mobile Extensions chapter to support the Avaya one-x mobile lite for iPhone client.
November 2010	Standard 05.01 and 05.02. This document is up-issued to support Communication Server 1000 Release 7.5.
May 2011	Standard 04.04. This document is up-issued to include the chapter Music on Hold, Set-based.
May 2011	Standard 04.03. This document is up-issued to include the chapter Music on Hold, Set-based.
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.

New in this release

Chapter 2: Customer service

Visit the Avaya Web site to access the complete range of services and support that Avaya provides. Go to www.avaya.com or go to one of the pages listed in the following sections.

Navigation

- [Getting technical documentation](#) on page 37
- [Getting product training](#) on page 37
- [Getting help from a distributor or reseller](#) on page 37
- [Getting technical support from the Avaya Web site](#) on page 38

Getting technical documentation

To download and print selected technical publications and release notes directly from the Internet, go to www.avaya.com/support.

Getting product training

Ongoing product training is available. For more information or to register, go to www.avaya.com/support. From this Web site, locate the Training link on the left-hand navigation pane.

Getting help from a distributor or reseller

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

Getting technical support from the Avaya Web site

The easiest and most effective way to get technical support for Avaya products is from the Avaya Technical Support Web site at www.avaya.com/support.

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 Mbit/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			
Analog Calling Line Identification	349	ACLI	25
• CLID on Analog Trunks for Hong Kong (A-CLID)			
Aries Digital Sets	170	ARIE	14

Package Name	Number	Mnemonic	Release
• 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 • Attendant Incoming Call Indicators • Attendant Interpositional Transfer • Attendant Lockout			
Attendant Overflow Position	56	AOP	1

Package Name	Number	Mnemonic	Release
• 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			
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) • 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			

Package Name	Number	Mnemonic	Release
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
• Automatic Call Distribution Call Waiting Thresholds (see also packages 40, and 131)			
• Automatic Call Distribution Least Call Queuing			
• Automatic Call Distribution Threshold Visual Indication (see also packages 40, and 131)			
Automatic Call Distribution, Priority Agent	116	PAGT	12
• Automatic Call Distribution Priority Agent			
Automatic Call Distribution, Timed Overflow Queuing	111	TOF	10
• ACD Timed Overflow			
• Automatic Gain Control Inhibit			
• Automatic Guard Detection			
• Automatic Hold			
Automatic ID of Outward Dialing	3	AIOD	1
Automatic Installation (Option 11 only)	200	AINS	16
• Automatic Installation			
Automatic Line Selection	72	LSEL	4
• Automatic Line Selection			

Package Name	Number	Mnemonic	Release
Automatic Number Identification Route Selection • Automatic Number Identification Route Selection	13	ANIR	1
Automatic Number Identification • Automatic Number Identification • Automatic Number Identification on DTI • Automatic Preselection of Prime Directory Number	12	ANI	1
Automatic Redial • Automatic Redial • Automatic Timed Reminders	304	ARDL	22
Automatic Wake-Up • Automatic Wake Up	102	AWU	10
Auxiliary Processor Link • Auxiliary Processor Link • Auxiliary Signaling • B34 Dynamic Loss Switching (see also packages 164 and 203)	109	APL	10
Background Terminal • Background Terminal Facility	99	BGD	10
Basic Alternate Route Selection • Network Alternate Route Selection/Basic Alternate Route Selection Enhancement – Local Termination (see also package 58)	57	BARS	1
Basic Authorization Code • Basic Authorization Code	25	BAUT	1
Basic Automatic Call Distribution • 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	40	BACD	1

Package Name	Number	Mnemonic	Release
• 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 • Basic Queuing	28	BQUE	1
Basic Rate Interface • Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 235)	216	BRI	18
Basic Routing • Basic Routing	14	BRTE	1
Boss Secretary Filtering (FFC activation) • Flexible Feature Code Boss Secretarial Filtering	198	FTCSF	15
BRI line application • 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	235	BRIL	18
Business Network Express • Business Network Express/EuroISDN Call Diversion • Business Network Express/EuroISDN Explicit Call Transfer • Business Network Express/Name and Private Number Display	367	BNE	25
Busy Tone Detection • China Phase II – Busy Tone Detection • Busy Tone Detection for Asia Pacific and CALA • Call Capacity Report	294	BTD	21
Call Center Transfer Connect • Call Center Transfer Connect	393	UUI	3.0

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

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

Package Name	Number	Mnemonic	Release
Centralized Attendant Services (Main) • Centralized Attendant Services - Main	26	CASM	1
Centralized Attendant Services (Remote) • Centralized Attendant Services – Remote • Centralized Multiple Line Emulation	27	CASR	1
Charge Account for CDR • Charge Account and Calling Party Number	23	CHG	1
Charge Account/Authorization Code • Charge Account/Authorization Code Base • Charge Display at End of Call (see also package 101)	24	CAB	1
China Attendant Monitor Package • 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)	285	CHINA	21
China Toll Package • China Phase II – Toll Call Loss Plan	292	CHTL	21
CLASS Calling Name Delivery • CLASS	333	CNAME	23
CLASS Calling Number Delivery • CLASS	332	CNUMB	23
Collect Call Blocking • Collect Call Blocking	290	CCB	21
Command Status Link • Command Status Link	77	CSL	8
Commonwealth of Independent States Multifrequency Shuttle Signaling • CIS Multifrequency Shuttle Signaling	326	CISMFS	23
Commonwealth of Independent States Trunks	221	CIST	21 24 24

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 • Commonwealth of Independent States Automatic Number Identification (ANI) Reception • Commonwealth of Independent States Toll Dial Tone Detection • Conference • Conference Warning Tone Enhancement for Italy			24
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			
Customer Controlled Routing	215	CCR	17
• Customer Controlled Routing • MFC Interworking with AML Based Applications (see also packages 128, and 214) • Dataport Hunting			
CP Pentium® Backplane for Intel® Machine	368	CPP_CNI	25
Deluxe Hold	71	DHLD	4
• Call Hold, Deluxe • Call Hold, Individual Hold Enhancement			
Departmental Listed Directory Number	76	DLDN	5

Package Name	Number	Mnemonic	Release
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			
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 • D-channel Handler Interface Expansion • Extension Three-Party Service • Loop Avoidance • Redirection • Route Optimization • Step Back on Congestion • Diversion			

Package Name	Number	Mnemonic	Release
- 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			
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			

Package Name	Number	Mnemonic	Release
- 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 Brand lining			
Emergency Services Access Calling Number Mapping	331	ESA_CLMP	23
• Emergency Services Access (See also packages 329 and 330)			
Emergency Services Access Supplementary	330	ESA_SUPP	23
• Emergency Services Access (See also packages 329 and 331)			
Emergency Services Access	329	ESA	23
• Emergency Services Access (See also packages 330 and 331) • End of Selection • End of Selection Busy • End-of-Dialing on Direct Inward/Outward Dialing Incoming Call Indicator Enhancement			
End-To-End Signaling	10	EES	1
• Attendant End-to-End Signaling • End-to-End Signaling			
Enhanced ACD Routing	214	EAR	17
• Enhanced Automatic Call Distribution Routing • MFC Interworking with AML Based Applications (see also packages 128, and 215)			
Enhanced Call Trace	215	ECT	18

Package Name	Number	Mnemonic	Release
• 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 • DPNSS1 Executive Intrusion	288	DPNSS_ES	21
Enhanced DPNSS1 Gateway • Enhanced DPNSS1 Gateway	284	DPNSS189I	20
Enhanced Hot Line • Hot Line • Network Intercom • Enhanced input/output buffering • Enhanced Maintenance (Patching)	70	HOT	4/10
Enhanced Music • Music, Enhanced	119	EMUS	12
Enhanced Night Service • Enhanced Night Service • Enhanced package printout • Equal Access Compliance	133	ENS	20
Euro ISDN Trunk - Network Side • EuroISDN Trunk - Network Side	309	MASTER	22
Euro ISDN • ISDN – Advice of Charge for EuroISDN • ISDN BRI and PRI Trunk Access for Europe (EuroISDN) • EURO ISDN Continuation	261	EURO	20
Euro Supplementary Service • EuroISDN Call Completion Supplementary Service	323	ETSI_SS	22
Executive Distinctive Ringing • Executive Distinctive Ringing	185	EDRG	16
FCC Compliance for DID Answer Supervision • Federal Communications Commission Compliance for DID Answer Supervision	223	FCC68	17

Package Name	Number	Mnemonic	Release
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
• 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	FNPN	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			

Package Name	Number	Mnemonic	Release
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
• 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_VTR K	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

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

Package Name	Number	Mnemonic	Release
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 • 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 Network Screen Activation and Flexible DN interactions • Intercept Treatment Enhancements			
Intercept Treatment	11	INTR	1
• Intercept Treatment			

Package Name	Number	Mnemonic	Release
Inter-Exchange Carrier	149	IEC	13
• Inter Exchange Carrier			
Internal CDR	108	ICDR	10
• Internal Call Detail Recording			
International 1.5/2.0 Mbit/s Gateway	167	GPRI	18
• Radio Paging			
• International Meridian 1			
International nB+D	255	INBD	20
• ISDN PRI Do 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)			

Package Name	Number	Mnemonic	Release
Japan Central Office Trunks	97	JPN	9
• Japan Central Office Trunk			
Japan Digital Multiplex Interface	136	JDMI	14
• Japan Digital Multiplex Interface			
Japan Telecommunication Technology Committee	335	JTTC	23
• Japan TTC Common Channel Signaling			
Japan Tone and Digit Switch	171	JTDS	14
• Japan Tone and Digit Switch			
Last Number Redial	90	LNR	8
• Last Number Redial			
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			
• 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
• Distinctive Ringing for Digital Telephones			
• M2317 Telephones			
• Flexible Voice/Data Terminal Number			

Package Name	Number	Mnemonic	Release
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 • Make Set Busy and Voice Call Override	210	MAID	17
Make Set Busy • Make Set Busy • Make Set Busy Improvement • Malicious Call Trace on Direct Inward Dialing	17	MSB	1
Malicious Call Trace • Enhanced Malicious Call Trace • Malicious Call Trace • Malicious Call Trace DN/TN Print • Malicious Call Trace Idle • Manual Line Service • Manual Service Recall to Attendant • Manual Signaling (Buzz) • Manual Trunk Service	107	MCT	10

Package Name	Number	Mnemonic	Release
MAT 5.0 • Meridian 1 Attendant Console Enhancements (see also package 76)	296	MAT	22
Meridian 1 Companion Option • Avaya Integrated DECT	240	MCMO	19
MCDN End to End Transparency	348	MEET	24
Meridian 1 Enhanced Conference, TDS and MFS • Meridian 1 Enhanced Conference, TDS and MFS	204	XCT0	15
Meridian 1 Fault Management • Alarm Management • Meridian 1 Initialization Prevention and Recovery	243	ALRM_FILTER	19
Meridian 1 Packet Handler • Meridian 1 Packet Handler	248	MPH	19
Meridian 1 Superloop Administration (LD 97) • 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	205	XCT1	15
Meridian 1 XPE • B34 Codec Static Loss Plan Downloading • B34 Dynamic Loss Switching (see also packages 131, and 164) • Extended Multifrequency Compelled Sender/Receiver • Extended Tone Detector and Global Parameters Download (see also package 205) • Intelligent Peripheral Equipment Software Support Enhancements	203	XPE	15
Meridian 911 • Meridian 911 Enhancements – Call Abandon • Meridian 911 Enhancements – MADN Display Coordination	224	M911	19
Meridian Hospitality Voice Service • Meridian Hospitality Voice Services	179	HVS	16

Package Name	Number	Mnemonic	Release
Meridian Link Modular Server • Meridian Link Enhancements	209	MLM	16
Meridian SL-1 ST Package • Meridian SL-1 ST Package	96	SLST	9
Message Intercept • Message Intercept	163	MINT	15
Message Waiting Center • Message Waiting Lamp Maintenance • Message Waiting Unconditional	46	MWC	1
Message Waiting Indication Interworking with DMS • Message Waiting Indication (MWI) Interworking	219	MWI	19
Mobile Extensions Modular Telephone Relocation	412	MOBX	5.50
Multifrequency Compelled Signaling • China Number 1 Signaling Trunk Enhancements (see also packages 49, 113, and 131) • China Number 1 Signaling – Active Feature Dial Tone (see also package 126) • China Number 1 Signaling – Audible Alarm (see also package 126) • China Number 1 Signaling – Vacant Number Announcement (see also package 126) • India Phase 2 • R2 Multifrequency Compelled Signaling (MFC) DID/DTMF DOD • R2 Multifrequency Compelled Signaling (MFC) Selective Route To Attendant • MFC Interworking with AML Based Applications (see also packages 214 and 215) • R2 Multifrequency Compelled Signaling Timer Control • Semi-Compelled MFC and Calling Name Identification Charges	128	MFC	9
Multifrequency Signaling for Socotel • Multifrequency Signaling for Socotel	135	MFE	10

Package Name	Number	Mnemonic	Release
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			
Multi-purpose Serial Data Link Single Terminal Access	228	MSDL STA	19
• Single Terminal Access			
Multi-purpose Serial Data Link	222	MSDL	18
• Multi-purpose Serial Data Link			
Multi-Site Mobility Networking	370	MSMN	25
Multi-User Login	242	MULTI_USE R	19
• Multi-User Login			
Music Broadcast	328	MUSBRD	23
• Music Broadcast			
Music	44	MUS	1
• Music			
Network Alternate Route Selection	58	NARS	1

Package Name	Number	Mnemonic	Release
• 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)			
Network Attendant Service	159	NAS	20
• Equi-distribution Network Attendant Service Routing (see also package 58) • Network Individual Do Not Disturb (See also packages 9 and 127).			
Network Authorization Code	63	NAUT	1
• Network Authorization Code			
Network Automatic Call Distribution	207	NACD	15
• Network Automatic Call Distribution			
Network Call Back Queuing	38	MCBQ	2
• Network Call Back Queuing			
Network Call Transfer	67	NXFR	3
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

Package Name	Number	Mnemonic	Release
• 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			
Avaya 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			

Package Name	Number	Mnemonic	Release
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 • 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			

Package Name	Number	Mnemonic	Release
Originator Routing Control	192	ORC_RVQ	18
• Network Drop Back Busy and Off-hook Queuing (see also package 62) • Remote Virtual Queuing • Out-of-Service Unit			
Outpulsing, asterisk (*) and octothorpe (#)	104	OPAO	
• Outpulsing of Asterisk "*" and Octothorpe "#"			
Overlap Signaling (M1 to M1 and M1 to 1TR6 CO)	184	OVLP	15
• Overlap Signaling • Overlay 45 Limited Repeats • Overlay Cache Memory • Override • Paging • Partial Dial Timing • PBX (500/2500) Telephones • Periodic Camp-on Tone • Periodic Clearing • Periodic Clearing Enhancement • Periodic Clearing on RAN, ACD, and Music			
Personal Call Assistant	398	PCA	3.0
• Personal Call Assistant			
Phantom TN	254	PHTN	20
• Phantom TNs • Position Busy with Call on Hold			
PPM/Message Registration	101	MR	10
• Advice of Charge Real-time Supplementary Services for NUMERIS and SWISSNET (see also package 131) • Advice of Charge – Charging Information and End of Call for NUMERIS Connectivity (see also package 131) • Message Registration • Periodic Pulse Metering • Predictive Dialing			
Pretranslation	92	PXLT	8

Package Name	Number	Mnemonic	Release
• Pretranslation • Preventing Reciprocal Call Forward			
Priority Network Override	389	PONW	25.40
• Network Break-in 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
Property Management System Interface	103	PMSI	10
• Property Management System Interface • Public Switched Data Service			
Pulsed E&M (Indonesia, French Colisée)	232	PEMD	18
• Pulsed E&M DTI2 Signaling			
Q Reference Signaling Point Interface	263	QSIG	20
• Integrated Services Digital Network QSIG Basic Call			
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			

Package Name	Number	Mnemonic	Release
Recorded Announcement Broadcast	327	RANBRD	23
• Recorded Announcement Broadcast			
Recorded Announcement	7	RAN	1
• Recorded Announcement			
Recorded Overflow Announcement	36	ROA	2
• Recorded Overflow Announcement			
• Recorded Telephone Dictation			
• Recovery of Misoperation on the Attendant Console			
• Recovery on Misoperation of Attendant Console			
• Reference Clock Switching			
• Reference Clock Switching (see also packages 75, 129, and 154)			
Remote IPE	286	REMOTE_I PE	
• Remote Intelligent Peripheral Equipment			
Remote Virtual Queuing	192	RVQ	18
• Network Drop Back Busy and Off-hook Queuing (see also package 62)			
• Remote Virtual Queuing			
Resident Debug	82	RSDB	9
• Restricted Call Transfer			
• Ring and Hold Lamp Status			
• Ringback Tone from Meridian 1 Enhancement			
Ringing Change Key	193	RCK	15
• Ringing Change Key			
Room Status	100	RMS	10
• Room Status			
Scheduled Access Restrictions	162	SAR	20
• Scheduled Access Restrictions			
• Secrecy Enhancement			
• Secretarial Filtering			
• Seizure Acknowledgment			

Package Name	Number	Mnemonic	Release
• Selectable Conferee Display and Disconnect • Selectable Directory Number Size			
Semi-Automatic Camp-On	181	SACP	15
• Attendant Blocking of Directory Number • Attendant Idle Extension Notification • Semi-Automatic Camp-On • Serial Port Expansion			
Series Call	191	SECL	15
• Series Call			
Set Relocation	53	SR	1
• Automatic Set Relocation • Short Buzz for Digital Telephones • Short Memory Test • Single Digit Access to Hotel Services			
Set-to-Set Messaging	380	STS	25
• Set-to-Set Messaging			
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			

Package Name	Number	Mnemonic	Release
Station Activity Records	251	SCDR	20
• Station Activity Records			
Station Camp-On	121	SCMP	20
• Station Camp-On			
Station Category Indication	80	SCI	7
• Station Category Indication			
Station Loop Preemption	106	SLP	10
Station Specific Authorization Codes	229	SSAU	19
• Station Specific Authorization Code			
• 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
Traffic Monitoring	168	TMON	

Package Name	Number	Mnemonic	Release
Trunk Anti-Tromboning	293	TAT	21
• Trunk Anti-Tromboning			
Trunk Barring	132	TBAR	20
• Trunk Barring			
Trunk Failure Monitor	182	TFM	15
• Trunk Failure Monitor			
• Trunk Failure Monitor Enhancement			
Trunk Hook Flash (Centrex)	157	THF	14
• Centrex Switchhook Flash			
• Italian Central Office Special Services (see also packages 129, and 131)			
• Trunk to Trunk Connections			
• Trunk Traffic Reporting Enhancement			
Trunk Verification from Station	110	TVS	9.32
• Trunk Verification from a Station			
• Uninterrupted Line Connection			
United Kingdom	190	UK	16
• Analog Private Network Signaling System (APNSS) (see also packages 122, 123, and 124)			
• UK Analogue Hardware Support			
Universal ISDN Gateways	283	UIGW	20
• Universal ISDN Gateway			
• Variable Flash Timing and Ground Button			
• Variable Guard Timing			
VIP Auto Wake Up	212	VAWU	17
• Hospitality Enhancements: V.I.P. Auto Wake Up			
Virtual Network Services	183	VNS	16
• Virtual Network Services			
• Virtual Network Services/Virtual Directory Number Expansion (see also package 58)			
• Voice Call			
Virtual Office	382	VIRTUAL_ OFFICE	25

Package Name	Number	Mnemonic	Release
• 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			
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: ICP Network Screen Activation, Flexible DN length

Contents

This section contains information on the following topics:

[Feature description](#) on page 75

[Operating parameters](#) on page 76

[Feature interactions](#) on page 77

[Feature packaging](#) on page 78

[Feature implementation](#) on page 78

[Feature operation](#) on page 79

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).

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.

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.

Feature interactions

The same interactions apply as for the ICP feature. 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, and Flexible DN:

- 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 1: LD 15](#) on page 79

For NWSA, enter up to 13 digits for ICDN and ECDN.

2. [Table 2: LD 15](#) on page 79

For FXDN, set the DN length and any padding digits at the ICDL and ICPD prompts.

Table 1: LD 15

Prompt	Response	Description
...		
TYPE:	ICP	Intercept computer update.
- ICP	YES	ICP is available.
...		
- ICMM	0-9	Message number.
- ICDN	0-13	Default internal DN.
- ECDN	0-13	Default external DN.

Table 2: 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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 5: In-Band Automatic Number Identification

Contents

This section contains information on the following topics:

[Feature description](#) on page 81

[Operating parameters](#) on page 82

[Feature interactions](#) on page 83

[Feature packaging](#) on page 86

[Feature implementation](#) on page 86

[Feature operation](#) on page 87

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 *Avaya 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 *Avaya 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 3: LD 16](#) on page 86
Identify the route as an In-Band Automatic Number Identification route.
2. [Table 4: LD 23](#) on page 87
Send the IANI messages across the Auxiliary Processor Link (APL).

Table 3: 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 4: 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 6: Incoming Call Indicator Enhancement

Contents

This section contains information on the following topics:

[Feature description](#) on page 89

[Operating parameters](#) on page 89

[Feature interactions](#) on page 90

[Feature packaging](#) on page 91

[Feature implementation](#) on page 91

[Feature operation](#) on page 91

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 5: 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 7: Incoming DID Digit Conversion

Contents

This section contains information on the following topics:

[Feature description](#) on page 93

[Operating parameters](#) on page 95

[Feature interactions](#) on page 96

[Feature packaging](#) on page 97

[Feature implementation](#) on page 97

[Feature operation](#) on page 99

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 6: LD 15](#) on page 98
Specify the maximum number of Incoming Digit Conversion trees allowed.
2. [Table 7: LD 49](#) on page 98
Create the IDC tables to convert incoming Direct Inward Dialing digits by specifying the IDC tree and customer numbers.
3. [Table 8: LD 16](#) on page 98
Enable the digit conversion for required DID trunk routes.

Table 6: 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 Avaya Communication Server (Avaya CS 1000E) system
- 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 7: 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 8: 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 Avaya CS 1000E system
IDC	YES	Use digit conversion for this route.

Prompt	Response	Description
- 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 8: Incoming Digit Conversion Enhancement

Contents

This section contains information on the following topics:

[Feature description](#) on page 101

[Operating parameters](#) on page 101

[Feature interactions](#) on page 102

[Feature packaging](#) on page 103

[Feature implementation](#) on page 103

[Feature operation](#) on page 103

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 93 in this document.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 9: Information Notification Service for Japan

Contents

This section contains information on the following topics:

[Feature description](#) on page 105

[Operating parameters](#) on page 108

[Feature interactions](#) on page 109

[Feature packaging](#) on page 110

[Feature implementation](#) on page 111

[Feature operation](#) on page 112

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 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 107 shows the operation of the INS-J feature.

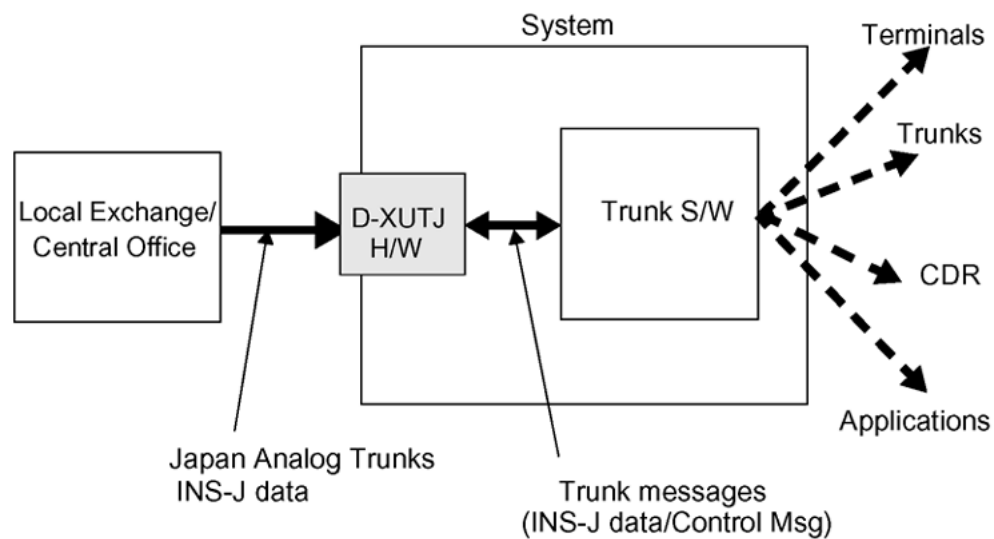


Figure 1: System with INS-J feature operating

[Figure 2: System composition for INS-J CLID delivery](#) on page 108 shows the system composition required for the INS-J CLID delivery.

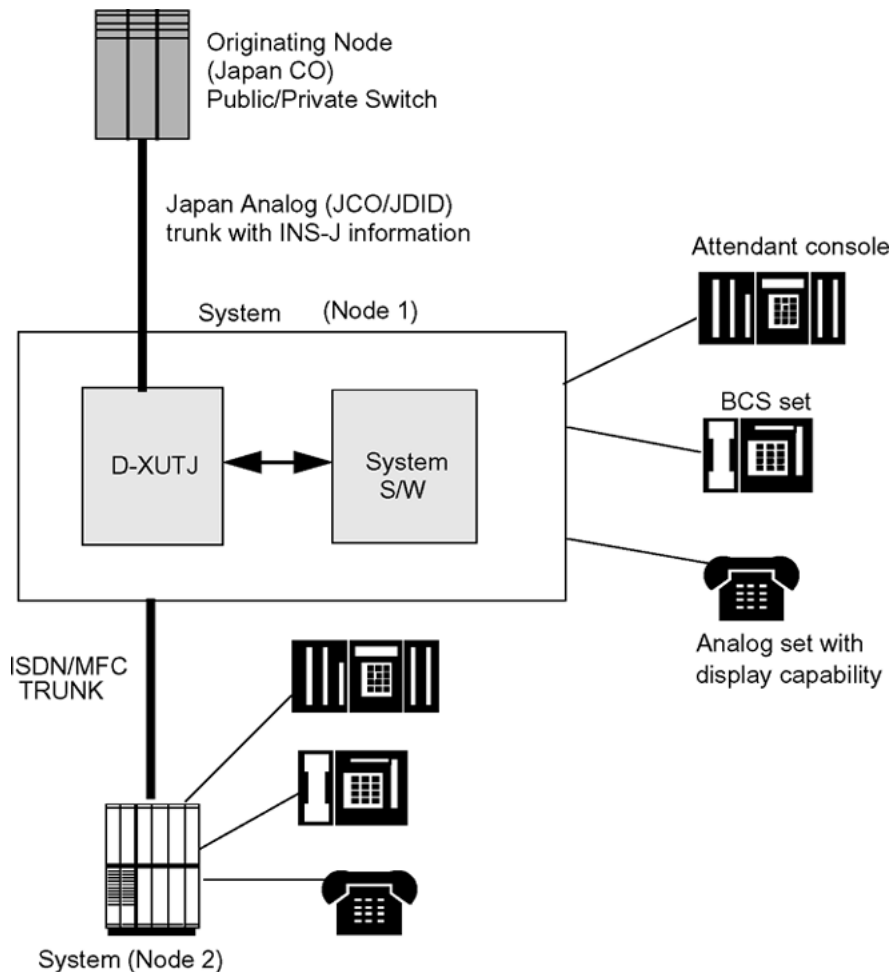


Figure 2: System composition for INS-J CLID delivery

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 9: LD 14](#) on page 111
Configure the Analog CLI Class of Service on a port-by-port basis
2. [Table 10: LD 16](#) on page 111
Configure the new ring validation timer

Table 9: 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 10: 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 10: Instant License

Contents

This section contains information on the following topics:

[Feature description](#) on page 113

[Feature interactions](#) on page 114

[Feature packaging](#) on page 115

[Feature implementation](#) on page 115

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

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 the following:

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

CP PIV systems store keycodes on the Fixed Media Device (FMD) or on the Removable Media Device (RMD). 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 process. If the new keycode does not show on the hard drive or the FMD, then the administrator must perform a new keycode installation, using LD 143.

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 two component parts:

1. Instant License parameter upgrade using a Removable Media Device (CP PIV systems)
2. Instant License parameter upgrade using HyperTerminal®

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 Avaya 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 118 if License limits are not increased or problems exist.

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 Avaya 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 118 if License limits are not increased or problems exist.

Chapter 11: Integrated Messaging System Link

Contents

This section contains information on the following topics:

[Feature description](#) on page 121

[Operating parameters](#) on page 122

[Feature interactions](#) on page 122

[Feature packaging](#) on page 122

[Feature implementation](#) on page 122

[Feature operation](#) on page 126

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 11: LD 17](#) on page 123
Add or change the link to a messaging system
2. [Table 12: LD 17](#) on page 123
Add or change the link to a messaging system
3. [Table 13: LD 15](#) on page 124
Add or change the IMS feature for a customer
4. [Table 14: LD 23](#) on page 125
Add or change ACD data for Integrated Messaging System Link feature
5. [Table 15: LD 11](#) on page 125
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 11: LD 17](#) on page 123

Table 11: 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 11: LD 17](#) on page 123

Table 12: 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 13: LD 15

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
OPT	(MCX) MCI	Message Center (excluded) included.
...		
TYPE	IMS	Integrated message service options.
- IMS	(NO) YES	(Do not) allow changes to the IMS feature.
- IMA	(NO) YES	IMS feature (is not) or is enabled.
- - APL	0-15	Port number of the link to the messaging system.
- UST	(NO) YES	User Status Update (UST) feature (is not) or is enabled.
- - APL	0-15	Port number of the link from UST to the messaging system.
- UMG	(NO) YES	User-to-User Messaging (UMG) feature (is not) enabled.

Prompt	Response	Description
-- APL	0-15	Port number of the link from UMG to the messaging system.

Table 14: 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 15: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit.
CLS	(IMD) IMA	This telephone (is not) is an IMS attendant.

Prompt	Response	Description
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 12: 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 *Avaya ISDN Primary Rate Interface Fundamentals*, NN43001-569 and *Avaya 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 *Avaya ISDN Basic Rate Interface Fundamentals*, NN43001-380 and *Avaya ISDN Basic Rate Interface Installation and Commissioning*, NN43001-318.

Chapter 13: Intelligent Peripheral, Equipment Completion

Contents

This section contains information on the following topics:

[Feature description](#) on page 129

[Operating parameters](#) on page 130

[Feature interactions](#) on page 130

[Feature packaging](#) on page 131

[Feature implementation](#) on page 131

[Feature operation](#) on page 135

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 limitations 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 16: LD 16](#) on page 132
Configure a RON/TRON signaling trunk route
2. [Table 17: LD 14](#) on page 132
Configure RON/TRON Signaling trunk

Table 16: 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 17: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit.
XTRK	XFEM	Extended Flexible E&M trunk card

Prompt	Response	Description
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 18: LD 16](#) on page 133
Configure L1 signaling on a XFEM TIE trunk route
2. [Table 19: LD 14](#) on page 134
Configure a L1 Signaling trunk

Use [Table 18: LD 16](#) on page 133 to configure L1 signaling on a XFEM TIE trunk route with Proceed to Send expected after an outgoing seize and answer supervision.

Table 18: 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
	ICF 0	Incoming Flash timer
	OFC 0	Outgoing Flash timer

Prompt	Response	Description
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 19: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, 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 20: LD 16](#) on page 135
Configure LDR signaling on XIDID trunk route
2. [Table 21: LD 14](#) on page 135
Configure LDR signaling on XIDID trunk

Table 20: 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 21: LD 14

Prompt	Response	Description
REQ	NEW	New
TYPE	TIE	TIE trunk data block
TN		Terminal number
	I s c u	Format for Large System, CS 1000E system, and MG 1000B where I = loop, s = shelf, c = card, 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 14: Intelligent Peripheral Equipment and Software Support Enhancements

Contents

This section contains information on the following topics:

[Feature description](#) on page 137

[Operating parameters](#) on page 138

[Feature interactions](#) on page 139

[Feature packaging](#) on page 139

[Feature implementation](#) on page 139

[Feature operation](#) on page 141

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.

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 22: LD 14](#) on page 140
Configure XFEM BPO trunk type signaling, and the Balance Impedance Adjustment on XDID trunk
2. [Table 23: LD 16](#) on page 140

Configure the Autoguard Repeat Prevention timer for the route

3. [Table 24: LD 97](#) on page 140

Configure the five XFALC timers to support downloading

Table 22: 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 23: 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 24: 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 15: Intercept Computer Interface

Contents

This section contains information on the following topics:

[Feature description](#) on page 143

[Operating parameters](#) on page 144

[Feature interactions](#) on page 145

[Feature packaging](#) on page 145

[Feature implementation](#) on page 146

[Feature operation](#) on page 149

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).

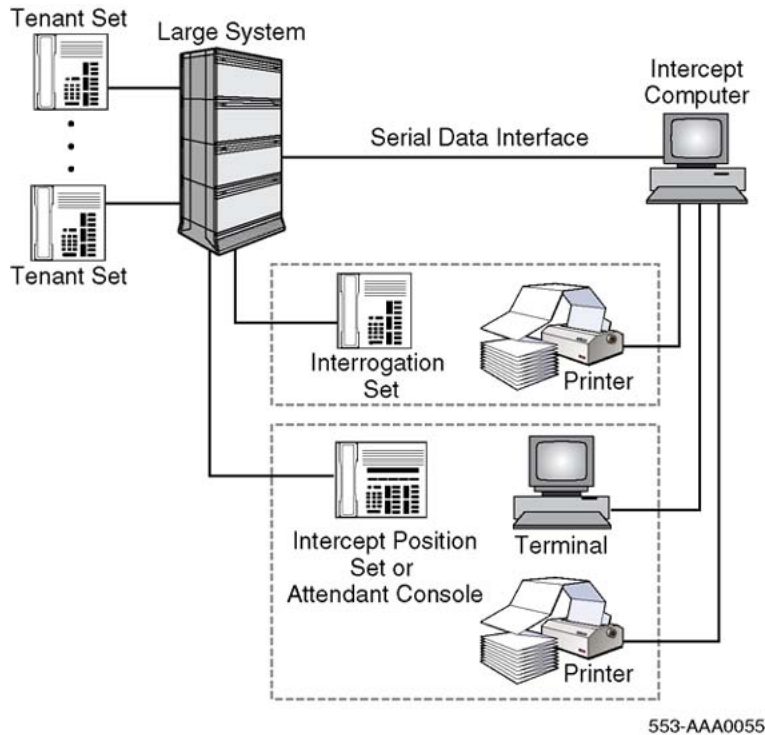


Figure 3: Intercept Computer Interface components

- 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 25: LD 17](#) on page 146
Configure the configuration record for ICP
2. [Table 26: LD 15](#) on page 147
Configure the customer data block for ICP
3. [Table 27: LD 10](#) on page 147
Create or modify the analog (500/2500-type) telephone data block for ICP
4. [Table 28: LD 11](#) on page 147
Create or modify the Meridian 1 proprietary telephone data block for ICP
5. [Table 29: LD 12](#) on page 148
Create or modify the attendant console data block for ICP
6. [Table 30: LD 23](#) on page 148
Modify the ACD/Message Center parameters for Incoming Call Indicators (ICIs)
7. [Table 31: LD 93](#) on page 148
Enable or modify the Multi-tenant Service feature

Table 25: 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 26: 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 27: 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 28: 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 29: 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 30: 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 31: 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)	Circular search.
	COM	One common printer for all consoles.

Feature operation

A terminal at the ICP displays a message stating why the tenant at the DN is absent. The person at the ICP can then store the caller 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 16: Intercept Computer Dial from Directory

Contents

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[Feature interactions](#) on page 152

[Feature packaging](#) on page 156

[Feature implementation](#) on page 157

[Feature operation](#) on page 158

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 32: LD 15](#) on page 157
Allow or deny an intercept attendant to dial an extension DN from the ICP
2. [Table 33: LD 15](#) on page 158
Set the switchhook flash time required when using package 131
3. [Table 34: LD 21](#) on page 158
- Print Intercept Computer Dial from directory system information

Table 32: 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 33: LD 15

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

Table 34: 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 17: Intercept Computer Enhancements

Contents

This section contains information on the following topics:

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[Operating parameters](#) on page 164

[Feature interactions](#) on page 164

[Feature packaging](#) on page 164

[Feature implementation](#) on page 164

[Feature operation](#) on page 166

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 35: LD 10](#) on page 165
Configure Intercept Computer Answering Machine Class of Service
- [Table 36: LD 15](#) on page 165
Configure internal and external call DNs for Intercept Transfer
- [Table 37: LD 93](#) on page 165
Configure internal and external call DNs for attendant console groups

Table 35: 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 36: 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 37: 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 18: Intercept Treatment

Contents

This section contains information on the following topics:

[Feature description](#) on page 167

[Operating parameters](#) on page 168

[Feature interactions](#) on page 169

[Feature packaging](#) on page 171

[Feature implementation](#) on page 171

[Feature operation](#) on page 172

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 38: Intercept Treatment for various types of calls](#) on page 168.

Table 38: 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 39: Intercept Treatment for BARS/NARS calls](#) on page 169 specifies the type of Intercept Treatments (INTR) available for BARS/NARS calls, and lists the intercept situations that are possible.

Table 39: 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 40: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
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.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 19: Intercept Treatment Enhancements

Contents

This section contains information on the following topics:

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[Feature interactions](#) on page 174

[Feature packaging](#) on page 174

[Feature implementation](#) on page 174

[Feature operation](#) on page 175

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 41: 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 20: International Meridian 1

Contents

This section contains information on the following topics:

[Feature description](#) on page 177

[Operating parameters](#) on page 178

[Feature interactions](#) on page 179

[Feature packaging](#) on page 179

[Feature implementation](#) on page 179

[Feature operation](#) on page 181

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
- 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 42: LD 17](#) on page 180
Configure the configuration record for 16-Button Dual-tone Multifrequency (DTMF) detection
2. [Table 43: LD 13](#) on page 180
Configure DTR, TDET, and DTD cards for this feature
3. [Table 44: LD 97](#) on page 180

Configure system parameters for Intelligent Peripheral Equipment in configuration record 2

Table 42: LD 17

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

Table 43: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit.

Table 44: LD 97

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

Prompt	Response	Description
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 21: Inventory Reporting

Contents

This section contains information on the following topics:

[Feature description](#) on page 183

[Operating parameters](#) on page 188

[Feature interactions](#) on page 194

[Feature packaging](#) on page 194

[Feature implementation](#) on page 195

[Feature operation](#) on page 195

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 45: Inventory Reporting generate commands](#) on page 183 for a list of commands and their descriptions.

Table 45: 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 47: Card inventory example for Meridian 1 Option 61C](#) on page 185 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 46: ID PROM information](#) on page 184 for more information.

Table 46: 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

See [Table 47: Card inventory example for Meridian 1 Option 61C](#) on page 185 for an example.

Table 47: 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

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 48: Inventory Reporting midnight routine commands](#) on page 186 for a list of commands and their descriptions.

Table 48: 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 49: Inventory Reporting print commands](#) on page 186 for a list of commands and their descriptions.

Table 49: 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 50: Inventory Reporting status responses](#) on page 187 for a list of status responses and their descriptions.

Table 50: 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 51: Card types are included in the Card Inventory file

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
COT	CO Trunk	NT5K93	AA, AB, BA, BB	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
CPP	System Utility Card	NT4N67	AA	Global
CPP	System Utility Transition Card	NT4N68	AA	Global
CPP	LED/LCD Display Panel	NT4N71	AA	Global
CPP	CCNI Card	NT4N65	AA	Global

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
CPU	68K Processor Card - Card Option CPU	NTAK14	AA, BA	North America
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
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
EXAL CC	Analog Line Card	NTRA08	AA, AB, BA	China
EXUT AP-1	Universal Trunk, Busy Tone detect Trunk, 400Hz	NTRA26	AA	Global
EXUT AP-2	Universal Trunk, Busy tone detect Trunk, 425Hz	NTRA26	BA	Global
EXUT C	Universal Trunk, Extended	NTRA10	AA, AB	China
EXUTJ	Universal Trunk	NT8D14	DA	Japan
EXUTJ	Universal Trunk, Extended	NT5D15	AA	Japan
FXNET	Fiber Extended Network	NTIP61	BA	Global
FXPEC	Fiber Extended Peripheral Equipment	NTIP62	CA	Global
IODUC	I/O Disk Unit w/ CD-ROM	NT5D61	AA, AB, 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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
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
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
XALC C	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
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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
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, LB, LC, LD, MA, MB, MC, NB, NC, PA, PB, PC, QA, QB, QC, SA, SB, TA, TB	Global
XFALC	Analog Line Card, Flexible High Voltage, Message'	NT5K96	MA, MB, NB, PB, SA, TA	Global
XFALC	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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
XFCOT	CO Trunk, Extended Flexible PPM, 8 unit	NT5K70	AA, AB	Global
XFCOT	CO Trunk, Extended Flexible	NTCK16	AA, BA, BC, BD, BE	Global
XFCOT	CO Trunk, Extended Flexible PPM	NTCK18	AA	Global
XFCOT	CO Trunk, Extended Flexible PPM	NTCK24	AA	Global
XFCOT	CO/FX/WATs Trunk	NT9C14	AA, BA	Global
XFEM	E & M Tie Trunk, Wire with recorded ann. & Paging	NT5K83	BA, BB, CA, CB	International
XFEM	E & M Tie Trunk, 4 Wire with recorded ann. & Paging	NT5K19	BA, BB,	International
XFEM	E & M Tie Trunk, Extended Flexible	NT5K50	AA	International
XFEM	E & M Tie Trunk, Extended Flexible	NT5K83	AA, AB, DA, DB, FA, GA, HA, KA, LA	International
XFEM	E & M Tie Trunk, Extended Flexible, 4 unit	NT5K72	AA	International
XFEM	E & M Trunk, Extended Flexible	NT5K19	AA, AB, AC	International
XMFC	Extended Multi-Frequency Compelled Sender Receiver	NT5K21	AA	International
XMFR	Extended MF Receiver	NTAG26	AA	International
XMLC	Message Waiting Line Card	NT5D49	AA	International
XMLC	Message Waiting Line Card	NT5D09	AA, BA, LA, PA	International
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

Card Mnemonic	Card Description	Product Engineering Code	Vintage	Market
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-Avaya (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
- Avaya 3901 Digital Deskphone
- Avaya 3902 Digital Deskphone
- Avaya 3904 Digital Deskphone
- Avaya 3905 Digital Deskphone

The Telephone Inventory file does not include ID PROM information for the following telephones:

- 500/2500 telephones
- Other digital telephones or any non-Avaya (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 22: IP Ad Hoc Conference

Contents

This section contains information on the following topics:

[Feature description](#) on page 197

[Operating parameters](#) on page 197

[Feature interactions](#) on page 198

[Feature packaging](#) on page 198

[Feature implementation](#) on page 199

[Feature operation](#) on page 202

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.

IP Media Services uses a floating license model to manage MAS licensing. In a floating license model, authorized users receive licenses from a licensing server. The licensing server provides a limited number of licenses that are shared by multiple users and applications on a system or network. When the license expires or is no longer required by a user or application, the licensing server reclaims the license for distribution to other authorized users or applications.

Before you can use this feature, you must configure IP Media Services. For information about configuring IP Media Services, see *Avaya 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 *Avaya 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 52: LD 17 - Create or modify an IP Ad Hoc Conference loop](#) on page 199
- [Table 53: LD 38 - Enable or disable IP Ad Hoc Conference loop](#) on page 200
- [Table 54: LD 117 - Print status of IP Ad Hoc Conference application](#) on page 200

Table 52: 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 53: 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 54: 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.

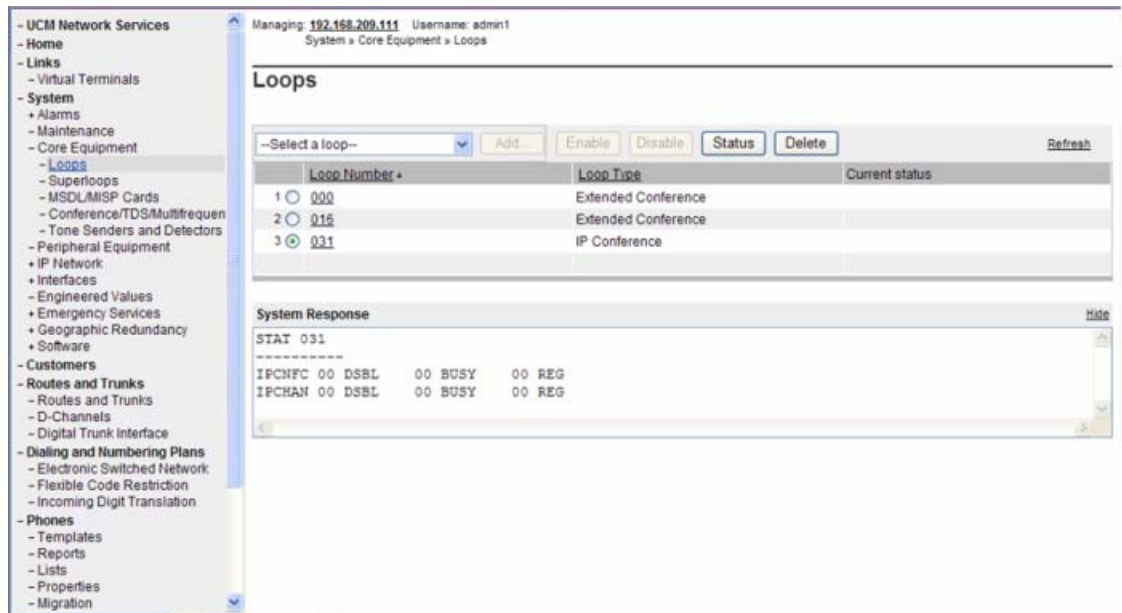


Figure 4: Element Manager Loops page

2. From the Select a loop list menu, select IP Conference.
- The IP Conference Loop Number Details page appears.

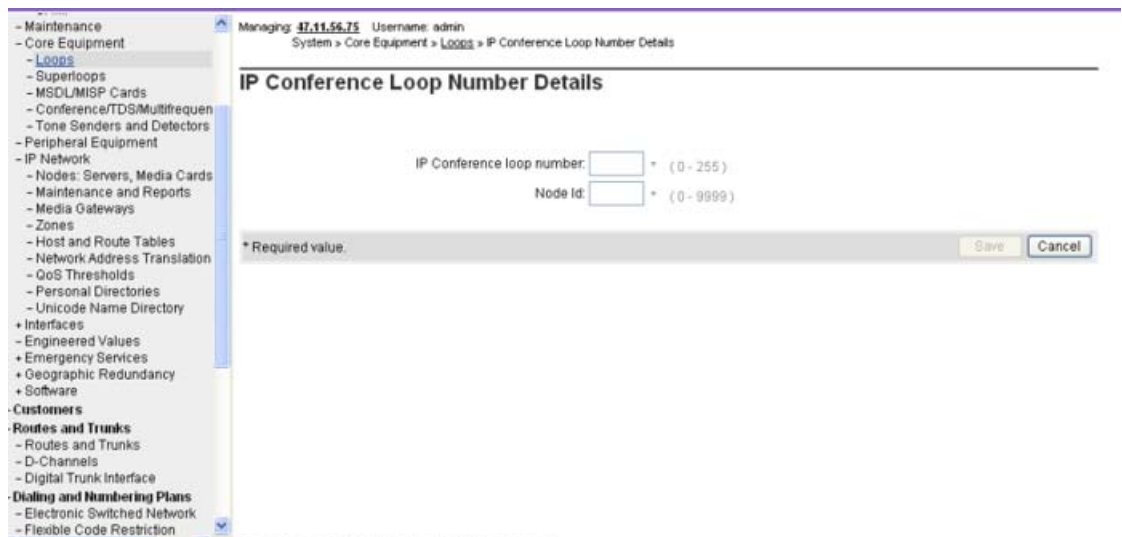


Figure 5: IP Conference Loop Number Details page

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 23: IP Attendant Console

Contents

This section contains information on the following topics:

[Feature description](#) on page 203

[Operating parameters](#) on page 210

[Feature interactions](#) on page 210

[Feature packaging](#) on page 210

[Feature implementation](#) on page 211

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 Avaya 2250 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 Avaya Communication Server 1000M (Avaya CS 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 Avaya 2250, 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 55: 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 56: 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 *Avaya 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 an Avaya 2250 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
- RIC1—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 *Avaya 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 57: 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.

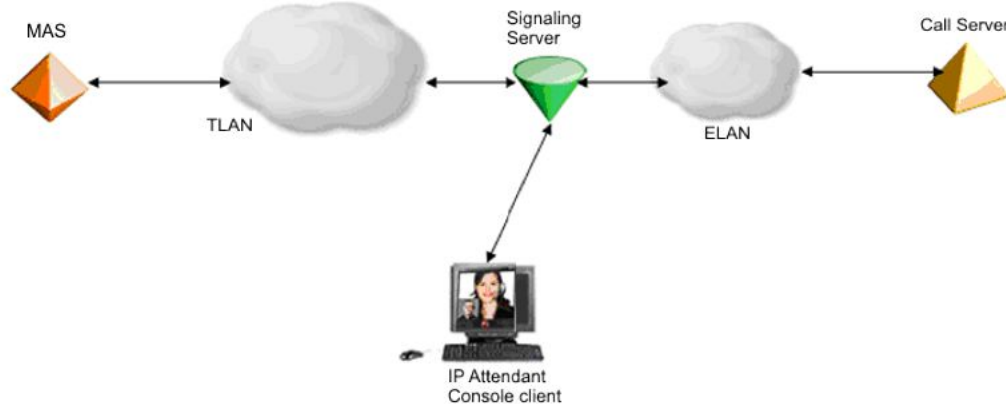


Figure 6: Direct deployment using 1 Call Server and 1 Signaling Server

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.

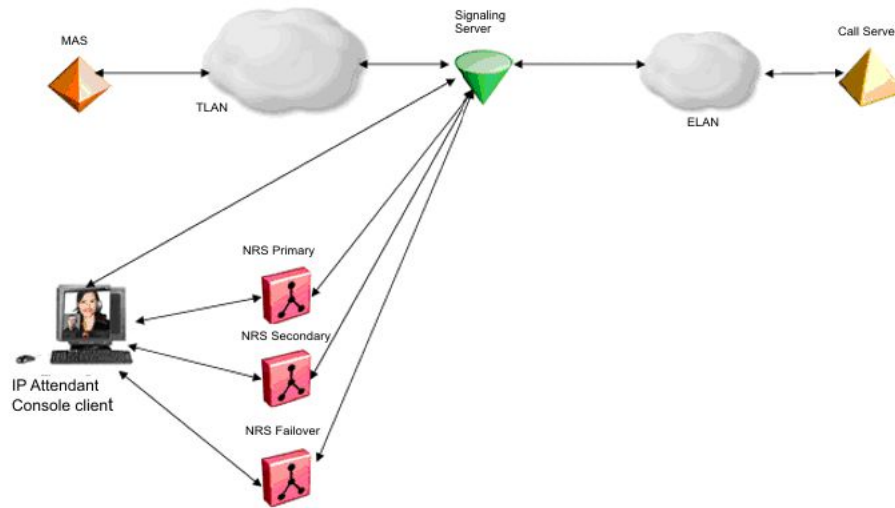


Figure 7: Redirect deployment using 1 NRS, 1 Signaling Server, and 1 Call Server

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 four additional SIP signaling 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 Avaya CS 1000 planning considerations, including traffic estimation calculations, see *Avaya 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 Media Services uses a floating license model to manage MAS licensing. In a floating license model, authorized users receive licenses from a licensing server. The licensing server provides a limited number of licenses that are shared by multiple users and applications on a system or network. When the license expires or is no longer required by a user or application, the licensing server reclaims the license for distribution to other authorized users or applications.

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 58: 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

Attendants	Estimated Traffic Load (Mbps)	Load Requirements
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
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 *Avaya Media Application Server Portal 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 *Avaya 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 Avaya 2250 Attendant 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 *Avaya 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 *Avaya 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 59: LD 12 - Add or modify an IP Attendant 3260 console](#) on page 212
- [Table 60: LD 20 - Print IP Attendant Console 3260 details](#) on page 212
- [Table 61: LD 22 - Print the total number of used and available IP Attendant Consoles](#) on page 213
- [Table 62: LD 81 - Print the number of IP Attendant 3260 consoles](#) on page 213
- [Table 63: LD 117 - Print IP Attendant information](#) on page 213

Table 59: 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 *Avaya Features and Services Fundamentals, Book 1, NN43001-106*.

Table 60: 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 61: 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 62: 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 63: 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 24: IP Music

Contents

This section contains information on the following topics:

[Feature description](#) on page 215

[Operating parameters](#) on page 215

[Feature interactions](#) on page 216

[Feature packaging](#) on page 216

[Feature implementation](#) on page 216

[Feature operation](#) on page 222

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 Avaya Communication Server (Avaya 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.

IP Media Services uses a floating license model to manage MAS licensing. In a floating license model, authorized users receive licenses from a licensing server. The licensing server provides a limited number of licenses that are shared by multiple users and applications on a system or network. When the license expires or is no longer required by a user or application, the licensing server reclaims the license for distribution to other authorized users or applications.

Before you can use this feature, you must configure IP Media Services. For information about configuring IP Media Services, see *Avaya Signaling Server 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 *Avaya 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 64: LD 16 - Create or change an IP Music route data block](#) on page 217
- [Table 65: LD 16 - Remove an IP Music route data block](#) on page 217
- [Table 66: LD 14 - Configure a trunk for an IP Music route data block](#) on page 218
- [Table 67: LD 11 - Assign MRT to a digital telephone](#) on page 218
- [Table 68: LD 11 - Enable Music on Hold for a digital telephone](#) on page 218
- [Table 69: LD 117 - Check status of IP Music application](#) on page 219

Table 64: 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 Avaya CS 1000E systems, the default is YES. CS 1000E systems only support broadcast trunks.

Table 65: LD 16 - Remove an IP Music route data block

Prompt	Response	Description
REQ	OUT	Remove data block

Prompt	Response	Description
TYPE	RDB	Route Data Block
CUST	xx	Customer number associated with the route
ROUT	x...x	Route number

Table 66: 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 67: 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 68: 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.
...		
CLS	(SBMA)/SBMD	Class of service, where:

Prompt	Response	Description
...		• SBMA = Set Based Music Allowed • SBMD = Set Based Music Denied

Table 69: 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.

Managing: 192.168.74.119 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, New Route Configuration

Customer 0, New Route Configuration

- Basic Configuration

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 0

Route number (ROUT): *

Designator field for trunk (DES):

Trunk type (TKTP):

Incoming and outgoing trunk (ICOG):

Access code for the trunk route (ACOD):

Calling number dialing plan (CNDP):

+ Basic Route Options

+ Network Options

+ General Options

+ Advanced Configurations

Save Cancel

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.

Managing: 192.168.74.119 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, New Route Configuration

Customer 0, New Route Configuration

- Basic Configuration

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 0

Route number (ROUT): *

Designator field for trunk (DES):

Trunk type (TKTP): IP Music trunk (IMUS)

Incoming and outgoing trunk (ICOG):

Access code for the trunk route (ACOD):

The route is for a virtual trunk route (VTRK): ☒

- Zone for codec selection and bandwidth management (ZONE): (0 - 255)

- Node ID of signaling server of this route (NODE): (0 - 9999)

Calling number dialing plan (CNDP): Unknown (UKWN)

+ Basic Route Options

+ Network Options

+ General Options

+ Advanced Configurations

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.

Managing: 192.168.74.119 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, Route 2 Property Configuration

Customer 0, Route 2 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE):

Customer number (CUST):

Route number (ROUT):

Designator field for trunk (DES):

Trunk type (TKTP):

Incoming and outgoing trunk (ICOG):

Access code for the trunk route (ACOD):

The route is for a virtual trunk route (VTRK): ☒

- Zone for codec selection and bandwidth management (ZONE): (0 - 255)

- Node ID of signaling server of this route (NODE): (0 - 9999)

Calling number dialing plan (CNDP):

+ Basic Route Options

+ Network Options

+ General Options

+ Advanced Configurations

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.

Managing: 192.168.74.110 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, Route 2, New Trunk Configuration

Customer 0, Route 2, New Trunk Configuration

- Basic Configuration

Input Description	Input Value
Multiple trunk input number (MTINPUT)	
Trunk data block (TYPE)	IMUS
Terminal Number (TN)	
Designator field for trunk (DES)	
Extended Trunk (XTRK)	IP Media Services trunk (IPMS)
Route number, Member number (RTMB)	
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)

+ Advanced Trunk Configurations

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 *Avaya 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 25: IP Network-wide Virtual Office

Contents

This section contains information on the following topics:

[Feature description](#) on page 223

[Operating parameters](#) on page 224

[Feature interactions](#) on page 225

[Feature packaging](#) on page 227

[Feature implementation](#) on page 228

[Feature operation](#) on page 230

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 Avaya 3900 Series Digital Deskphones, see the Avaya 3900 Series Digital Deskphones chapter in *Features and Services Fundamentals—Book 1 of 6, NN43001–106*. For more information on Phantom TNs and Virtual TNs, see [Incremental Software Management](#) on page 678.

Upon login, 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 *Avaya Software Input Output - Administration, NN43001-611*.

Virtual Office is supported for the IP Phone 2001, IP Phone 2002, IP Phone 2004, Avaya 2007 IP Deskphone, Avaya 2033 IP Conference Phone, Avaya 1110 IP Deskphone, Avaya 1120E IP Deskphone, Avaya 1140E IP Deskphone, Avaya 1150E IP Deskphone, Avaya 2050 IP Softphone, MVC 2050, and WLAN Handset 2210/2211/2212. The IP Phone 2002, IP Phone 2004, Avaya 2050 IP Softphone, and Mobile Voice Client (MVC) 2050, IP Phone support the IP Network-wide Virtual Office feature. For more information, see *Avaya 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 *Avaya 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
- Avaya 2050 IP Softphone 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, Avaya Communication Server 1000 (Avaya CS 1000) Release 5.0 does not support the Avaya 1200 Series IP Deskphones, therefore an Avaya 1200 Series IP Deskphones Virtual Office user cannot perform an IP Network-wide Virtual Office log on to a TN that is located on an Avaya CS 1000 Release 5.0.

Multiple Appearance Directory Number

If logging on using an 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 logging on using the single appearance DN, the user receives the assigned keymap and feature key list including the MADN.

If there is no matching SCPW, a `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 678 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.

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 70: LD 15 – Configuring Home Location Code \(for UDP only\)](#) on page 228
2. [Table 71: LD 11 – Configure IP Network-wide Virtual Office](#) on page 228
3. [Table 72: LD 86 – Configure Electronic Switched Network](#) on page 229
4. [Table 73: LD 87 – Configure the Local and Distant Steering Codes \(for CDP only\)](#) on page 229
5. [Table 74: LD 90 – Configuring the Access Code \(for UDP only\)](#) on page 229

Table 70: 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.
...		
- 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 71: 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.

Prompt	Response	Description
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 72: 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 73: 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.
FEAT	CDP	Coordinated Dialing Plan (CDP).
TYPE	LSC	Local Steering Code (LSC).
LSC	x...x	LSC.
...		
DSC	x...x	Distant Steering Code (DSC)

Table 74: 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 Fundamentals*, (NN43001-130). For more information about the H.323 ID for the Signaling Server in Element Manager, see *Avaya 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 *Avaya 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 230
- [IP Network-wide Virtual Office connection](#) on page 235
- [IP Network-wide Virtual Office logout](#) on page 236

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

- An IP Network-wide Virtual Office Login from an Avaya 2050 IP Softphone to an IP Phone 2001 is blocked.
- An IP Network-wide Virtual Office Login from an Avaya 2050 IP Softphone 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 Avaya 2050 IP Softphone, or the MVC 2050 are not accessible when logged in from an IP Phone 2002.

Use [Performing IP Network-wide Virtual Office login](#) on page 231 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, Avaya 2033 IP Conference Phone, 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 75: System messages](#) on page 233 for system messages, potential causes, and available actions.

Table 75: 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.

Message	Cause	Action
	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 and Avaya 2050 IP Softphone TN from an IP Phone 2002.	Go to an IP Phone 2004 and Avaya 2050 IP Softphone 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).
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 Avaya 2033 IP Conference Phone from an Avaya 2050 IP Softphone. Attempted Login to an IP Phone 2002 or IP Phone 2004 with CLS HFD from an Avaya 2050 IP Softphone.	Go to an IP Phone 2001, Avaya 2033 IP Conference Phone, 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	Update firmware.

Message	Cause	Action
	with that provided by the local TPS node.	
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 CS 1000 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 236).
- 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 236)

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 236 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, Avaya 2033 IP Conference Phone 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 *Avaya Software Input Output - Administration, NN43001-611*.

Chapter 26: IP Phone Audio Message Waiting Indication

Contents

This section contains information about the following topics:

- [Feature description](#) on page 239
- [Operating parameters](#) on page 239
- [Feature interactions](#) on page 240
- [Feature packaging](#) on page 240
- [Feature implementation](#) on page 240
- [Feature operation](#) on page 241

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 241.

System: EM on und-tpsf-b
Phone Type: 2050PC
Sync Status: TRN

General Properties | Features | Keys | UserFields |

General Properties

Customer Number: 0 *

Terminal Number: 086 0 00 03

Designation: 2050 * (1-6 characters)

Zone: 000 *

Key Expansion Modules: 0

Features

Feature	Description	Value
MWA	Message Waiting at Message Service	Denied
MWTA	Message waiting tone allowed	Denied
NAJA	No-Answer Indication	Denied
NAMA	Present/Restrict Calling Name	Allowed

Figure 12: MWTa on Phone Details Web page

Feature operation

No specific operating procedures are required to use this feature.

Chapter 27: IP Phone Disable Mute function

Contents

This section contains information about the following topics:

- [Feature description](#) on page 243
- [Operating parameters](#) on page 244
- [Feature interactions](#) on page 244
- [Feature packaging](#) on page 244
- [Feature implementation](#) on page 245
- [Feature operation](#) on page 246

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
- Avaya 2007 IP Deskphone
- Avaya 2033 IP Conference Phone
- Avaya 2050 IP Softphone
- Mobile Voice Client 2050
- Avaya 1120E IP Deskphone
- Avaya 1140E IP Deskphone
- Avaya 1150E IP Deskphone
- Avaya 1165E IP Deskphone
- Avaya 1210 IP Deskphone
- Avaya 1220 IP Deskphone
- Avaya 1230 IP Deskphone

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 245.

The screenshot shows the 'Phone Details' page in Element Manager. The left sidebar contains a navigation tree with categories like UCM Network Services, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, Tools, and Security. The 'Phones' category is selected. The main content area is titled 'Phone Details' and includes a phone icon, system information (System: EM on und-tpsf-6, Phone Type: 1120, Sync Status: NEW), and tabs for General Properties, Features, Jobs, and User Fields. The 'General Properties' tab is active, showing fields for Customer Number, Terminal Number, Designation, Zone, and Key Expansion Modules. Below this, the 'Features' section contains a table with columns for Feature, Description, and Value.

Feature	Description	Value
MSEC	Media Security Encryption	Media Security Never(MSNV)
MSIA	Make Set Busy Improvement	Denied
MTA	Maintenance Set	Denied
MUTA	Mute allowed	Allowed

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 28: IP Phone Password Protection for Language and Feature Key Labels

Contents

This section contains information about the following topics:

- [Feature description](#) on page 247
- [Operating parameters](#) on page 247
- [Feature interactions](#) on page 248
- [Feature packaging](#) on page 248
- [Feature implementation](#) on page 248
- [Feature operation](#) on page 249

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 248
2. [Define SCPW in Element Manager](#) on page 249

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 249.

Controlled Class of Service Web page

Managing: 192.167.166.3
Customers > Customer 00 > Edit > Controlled Class of Service

Controlled Class of Service

Restricted service:

Enhanced Level 1:
Customer defined first level of restriction

Enhanced Level 2:
Customer defined second level of restriction

Network wide electronic lock: (0-99)

Controlled Network Class of Service. Please refer help file to map values to Class of Service.

Electronic lock on private lines: ☐

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 29: 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](#) on page 251
- [Operating parameters](#) on page 253
- [Feature interactions](#) on page 253
- [Feature packaging](#) on page 253
- [Feature implementation](#) on page 253
- [Feature operation](#) on page 254

Feature description

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

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 252.

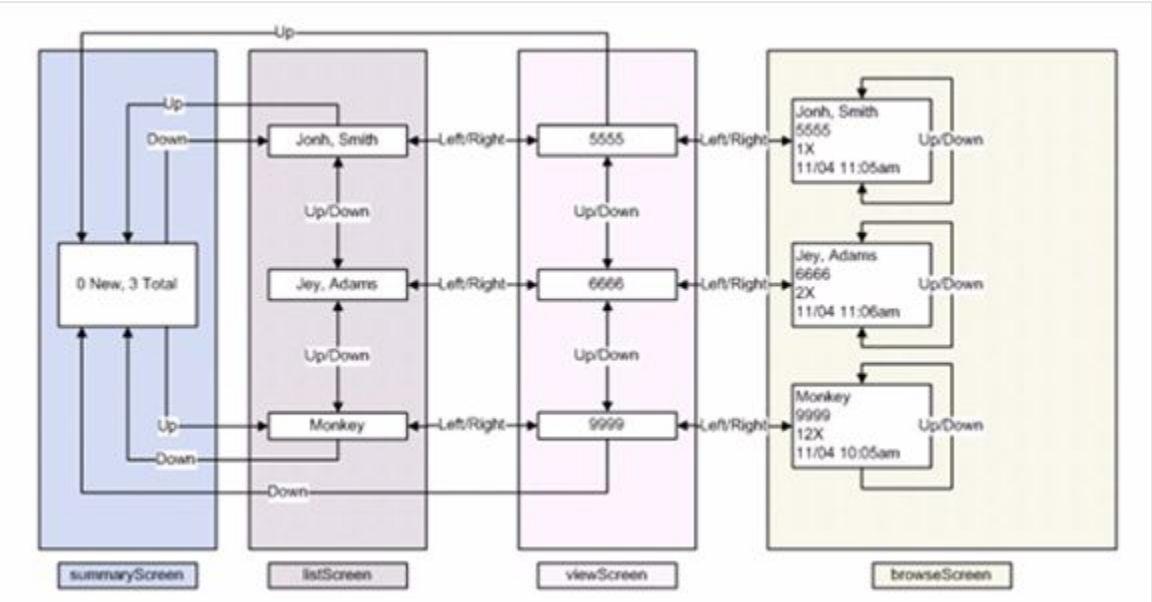


Figure 15: Single-line-display IP Phones

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 252.

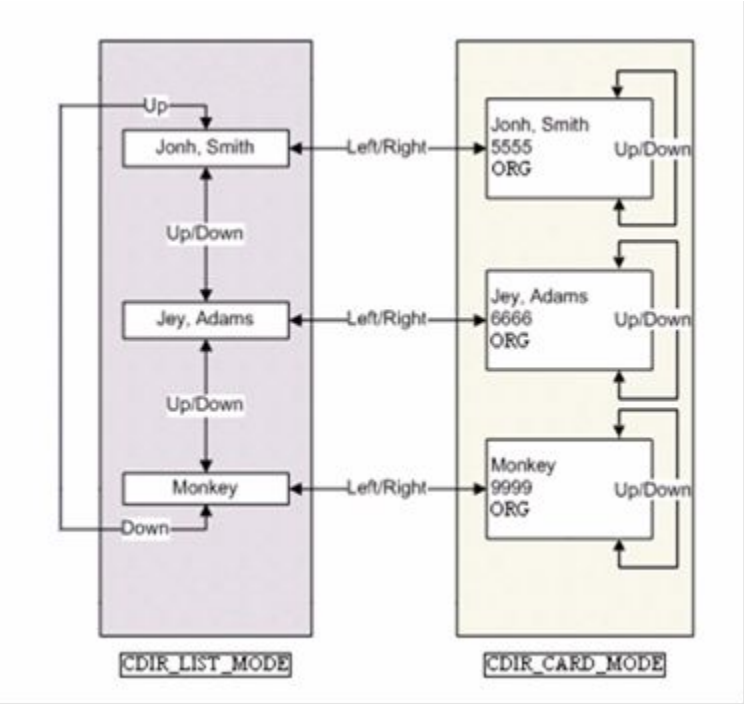


Figure 16: Corporate Directory

Operating parameters

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

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 76: 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 30: IP Recorded Announcements

Contents

This section contains information on the following topics:

[Feature description](#) on page 255

[Operating parameters](#) on page 256

[Feature interactions](#) on page 256

[Feature packaging](#) on page 256

[Feature implementation](#) on page 257

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 *Avaya Features and Services Fundamentals, Book 5, NN43001-106*.

Operating parameters

Before you can use this feature, you must configure IP Media Services. For information about configuring IP Media Services, see *Avaya Signaling Server 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.

IP Media Services uses a floating license model to manage MAS licensing. In a floating license model, authorized users receive licenses from a licensing server. The licensing server provides a limited number of licenses that are shared by multiple users and applications on a system or network. When the license expires or is no longer required by a user or application, the licensing server reclaims the license for distribution to other authorized users or applications.

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

Feature interactions

For more information about feature interactions, see the entry for Recorded Announcements in *Avaya 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 77: LD 16 - Create or modify IP RAN route data block](#) on page 257
- [Table 78: LD 14 - Configure a trunk for an IP RAN route data block](#) on page 258

Table 77: 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.

Prompt	Response	Description
POST	ATT	Call is routed to attendant after specified number of repetitions (applies to Direct Inward Dial [DID] calls on Intercept).
	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 Avaya Communication Server 1000E (Avaya CS 1000E) systems, the default is YES. Avaya 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 78: 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 79: 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.

Managing: 192.168.74.110 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, New Route Configuration

Customer 0, New Route Configuration

- Basic Configuration

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 0

Route number (ROUT):

Designator field for trunk (DES):

Trunk type (TKTP): IP Recorded Announcement trunk (IRAN)

Access code for the trunk route (ACOD):

The route is for a virtual trunk route (VTRK): ☒

- Zone for codec selection and bandwidth management (ZONE): (0 - 255)

- Node ID of signaling server of this route (NODE): (0 - 9999)

Recording device for RAN trunks (RTYP): Media Application Server (MAS)

Repetitions of recorded announcements (REP):

RAN Post announcement treatment (POST):

Start arrangement (STRT):

Calling number dialing plan (CNDP): Unknown (UKWN)

+ Basic Route Options

- Network Options

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.

Managing: 192.168.74.110 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, Route 2 Property Configuration

Customer 0, Route 2 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 00

Route number (ROUT): 2

Designator field for trunk (DES): IRAN

Trunk type (TKTP): IRAN

Access code for the trunk route (ACOD): 2345

The route is for a virtual trunk route (VTRK): ☒

- Zone for codec selection and bandwidth management (ZONE): 00000 (0 - 255)

- Node ID of signaling server of this route (NODE): 4040 (0 - 9999)

Recording device for RAN trunks (RTYP): Media Application Server (MAS)

Repetitions of recorded announcements (REP): 2

RAN Post announcement treatment (POST): Disconnect after maximum repetitions (DIS)

Start arrangement (STRT): Immediately connect call to recording (IMM)

Calling number dialing plan (CNDP): Unknown (UKWN)

+ Basic Route Options

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.

Managing: 192.168.74.110 Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, Route 2, New Trunk Configuration

Customer 0, Route 2, New Trunk Configuration

- Basic Configuration

Input Description	Input Value
Multiple trunk input number (MTINPUT)	Range: 1 - 3700
Trunk data block (TYPE)	IRAN
Terminal Number (TN)	*
Designator field for trunk (DES)	
Extended Trunk (XTRK)	IP Media Services trunk (IPMS)
Route number, Member number (RTMB)	*
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)

- Advanced Trunk Configurations

Maximum broadcast connections allowed (CONN)	4

Save Cancel

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 *Avaya Element Manager System Reference - Administration, NN43001-632*.

Chapter 31: IP Tone

Contents

This section contains information on the following topics:

[Feature description](#) on page 263

[Operating parameters](#) on page 264

[Feature interactions](#) on page 264

[Feature packaging](#) on page 264

[Feature implementation](#) on page 264

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.

IP Media Services uses a floating license model to manage MAS licensing. In a floating license model, authorized users receive licenses from a licensing server. The licensing server provides a limited number of licenses that are shared by multiple users and applications on a system or network. When the license expires or is no longer required by a user or application, the licensing server reclaims the license for distribution to other authorized users or applications.

Before you can use this feature, you must configure IP Media Services. For information about configuring IP Media Services, see *Avaya Signaling Server IP Line Applications Fundamentals*, NN43001–125.

Feature interactions

For information about feature interactions, see the entry for Tones and Cadences in *Avaya 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 80: LD 17 - Configure a virtual IP Tone loop](#) on page 265
- [Table 81: LD 34 - Enable or disable IP Tone loop](#) on page 265
- [Table 82: LD 117 - Print status of IP Tone application](#) on page 266

Table 80: 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 81: 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 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

Command	Description
	- 30 = Maximum number of IP tones are registered and available for use

Table 82: 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.

The screenshot shows the 'IP TDS Loop Number Details' page in the Element Manager. On the left is a navigation tree under 'UCM Network Services' with categories like Home, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans, and Phones. The 'System' category is expanded, showing 'Core Equipment' and 'Loops'. The 'Loops' sub-category is selected. The main content area has a header 'Managing: 47.11.56.75 Username: admin' and a breadcrumb 'System > Core Equipment > Loops > IP TDS Loop Number Details'. Below the header is the title 'IP TDS Loop Number Details'. There are two input fields: 'IP TDS loop number:' with a range of '(0 - 255)' and 'Node Id:' with a range of '(0 - 9999)'. Both fields are marked with an asterisk. A grey bar at the bottom of the form area contains the text '* Required value.'

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 32: Italian Central Office Special Services

Contents

This section contains information on the following topics:

[Feature description](#) on page 269

[Operating parameters](#) on page 269

[Feature interactions](#) on page 270

[Feature packaging](#) on page 271

[Feature implementation](#) on page 271

[Feature operation](#) on page 272

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.

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 83: 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 RIs and Hold keys may be activated during a call to a special "1xx" service.

Chapter 33: Italian Periodic Pulse Metering

Contents

This section contains information on the following topics:

[Feature description](#) on page 273

[Operating parameters](#) on page 274

[Feature interactions](#) on page 274

[Feature packaging](#) on page 274

[Feature implementation](#) on page 275

[Feature operation](#) on page 275

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

The feature does not work on the following DTI2 cards: QPC915A, QPC915B, QPC536A, QPC536B, and QPC536C. 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 84: 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 34: 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 35: Last Number Redial

Contents

This section contains information on the following topics:

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[Operating parameters](#) on page 279

[Feature interactions](#) on page 280

[Feature packaging](#) on page 284

[Feature implementation](#) on page 284

[Feature operation](#) on page 285

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 85: LD 15](#) on page 284
Enable or disable LNR for a customer.
2. [Table 86: LD 10](#) on page 284
Add or change LNR for analog (500/2500-type) telephones.
3. [Table 87: LD 11](#) on page 285
Add or change LNR for Meridian 1 proprietary telephones.

Table 85: LD 15

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

Table 86: LD 10

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	500	Telephone type
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
CLS	(LND) LNA	LNR (denied) allowed
LNRS	4-(16)-31	LNR size

Table 87: 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
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 36: Limited Access to Overlays

Contents

This section contains information on the following topics:

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[Operating parameters](#) on page 289

[Feature interactions](#) on page 290

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[Feature operation](#) on page 293

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 *Avaya 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 88: LD 17](#) on page 291
Define LAPW options and passwords.
2. [Table 89: LD 17](#) on page 292
Change user LAPW password (user must log in using current LAPW).
3. [Table 90: LD 22](#) on page 292
Check options available for LAPW passwords (administrator).
4. [Table 91: LD 22](#) on page 293
Print options for LAPW password (user).
5. [Table 92: LD 22](#) on page 293
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 88: 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
- LAPW	<cr>	Print Only Class of Service (denied) allowed. Printing Speed Call lists (allowed) denied. 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 89: 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 90: 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 91: 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 92: 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 37: Limited Access to Overlays Password Enhancement

Contents

This section contains information on the following topics:

[Feature description](#) on page 295

[Operating parameters](#) on page 296

[Feature interactions](#) on page 296

[Feature packaging](#) on page 296

[Feature implementation](#) on page 296

[Feature operation](#) on page 297

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 93: 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		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 38: Line and Trunk Cards

Contents

This section contains information on the following topics:

[Feature description](#) on page 299

[Operating parameters](#) on page 301

[Feature interactions](#) on page 301

[Feature packaging](#) on page 301

[Feature implementation](#) on page 301

[Feature operation](#) on page 301

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 *Avaya 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 *Avaya 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 39: Line Load Control

Contents

This section contains information on the following topics:

[Feature description](#) on page 303

[Operating parameters](#) on page 305

[Feature interactions](#) on page 305

[Feature packaging](#) on page 305

[Feature implementation](#) on page 305

[Feature operation](#) on page 307

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 304 shows 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.

Station Class of Service				
	LLCN	LLC1	LLC2	LLC3
T	Stations immune to LCC	LLC1, LLC2, and LLC3		
S		LLC1 and LLC2		No control
F		LLC1	No control	No control
OFF		No control (LLC off)		

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 94: LD 10](#) on page 306

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

2. [Table 95: LD 11](#) on page 306

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

3. [Table 96: LD 2](#) on page 306

Set Line Load Control levels.

Table 94: LD 10

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

Table 95: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit.
CLS	(LLCN)	LLC not enabled (default)
	LLC1	LLC class 1
	LLC2	LLC class 2
	LLC3	LLC class 3

Table 96: LD 2

Prompt	Response	Description
SCTL	x aaa	Set blocking probability 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

Prompt	Response	Description
TLLC		x = F (LLC, level F) x = S (LLC, level S) x = T (LLC, level T) OFF (deactivate LLC) Print blocking probability and current active LLC level.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 40: Line Lockout

Contents

This section contains information on the following topics:

[Feature description](#) on page 309

[Operating parameters](#) on page 310

[Feature interactions](#) on page 310

[Feature packaging](#) on page 312

[Feature implementation](#) on page 312

[Feature operation](#) on page 313

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 97: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
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 41: Listed Directory Numbers

Contents

This section contains information on the following topics:

[Feature description](#) on page 315

[Operating parameters](#) on page 315

[Feature interactions](#) on page 316

[Feature packaging](#) on page 316

[Feature implementation](#) on page 316

[Feature operation](#) on page 317

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 98: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
- 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 42: Listed Directory Numbers, Network Wide

Contents

This section contains information on the following topics:

[Feature description](#) on page 319

[Operating parameters](#) on page 320

[Feature interactions](#) on page 320

[Feature packaging](#) on page 321

[Feature implementation](#) on page 322

[Feature operation](#) on page 324

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 99: LD 15](#) on page 322
Activate Network Wide LDN in CDB.
2. [Table 100: LD 15](#) on page 323
Add or change LDN keys.
3. [Table 101: LD 93](#) on page 323
Add or change LDN keys in CPG.

Table 99: 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 100: 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 101: 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 43: Lockout, DID Second Degree Busy, and MFE Signaling Treatments

Contents

This section contains information on the following topics:

[Feature description](#) on page 325

[Operating parameters](#) on page 326

[Feature interactions](#) on page 326

[Feature packaging](#) on page 326

[Feature implementation](#) on page 327

[Feature operation](#) on page 327

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 DNSs.

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 102: 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 44: Loop Start Supervisory Trunks

Contents

This section contains information on the following topics:

[Feature description](#) on page 329

[Operating parameters](#) on page 330

[Feature interactions](#) on page 330

[Feature packaging](#) on page 331

[Feature implementation](#) on page 331

[Feature operation](#) on page 332

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, Avaya 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 103: LD 14](#) on page 332

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

2. [Table 104: LD 16](#) on page 332

Table 103: 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 104: 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 45: Loop Start Supervisory Trunks (Incoming Calls)

Contents

This section contains information on the following topics:

[Feature description](#) on page 333

[Operating parameters](#) on page 334

[Feature interactions](#) on page 334

[Feature packaging](#) on page 335

[Feature implementation](#) on page 335

[Feature operation](#) on page 335

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 be used for these services.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Table 105: 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 46: Loopback on Public Exchange or Central Office Trunks

Contents

This section contains information on the following topics:

[Feature description](#) on page 337

[Operating parameters](#) on page 337

[Feature interactions](#) on page 338

[Feature packaging](#) on page 338

[Feature implementation](#) on page 338

[Feature operation](#) on page 338

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 47: Make Set Busy and Voice Call Override

Contents

This section contains information on the following topics:

[Feature description](#) on page 339

[Operating parameters](#) on page 339

[Operating parameters](#) on page 339

[Feature interactions](#) on page 340

[Feature packaging](#) on page 340

[Feature implementation](#) on page 340

[Feature operation](#) on page 341

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 106: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
...		

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 48: Make Set Busy

Contents

This section contains information on the following topics:

[Feature description](#) on page 343

[Operating parameters](#) on page 344

[Feature interactions](#) on page 344

[Feature packaging](#) on page 348

[Feature implementation](#) on page 348

[Feature operation](#) on page 349

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 *Avaya 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 107: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, 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.

Make Set Busy

Chapter 49: Make Set Busy Improvement

Contents

This section contains information on the following topics:

[Feature description](#) on page 351

[Operating parameters](#) on page 352

[Feature interactions](#) on page 352

[Feature packaging](#) on page 352

[Feature implementation](#) on page 353

[Feature operation](#) on page 354

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 108: LD 11](#) on page 353

Activate the MSBI feature and define Primary DN telephone (boss) with non-ringing DN key.

2. [Table 109: LD 11](#) on page 354

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

Table 108: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit.
...	...	
CLS	MSIA	Allow Make Set Busy Improvement feature. (MSID) = Deny Make Set Busy Improvement feature.
...	...	
KEY		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.
...	xx SCN yyyy xx MCN yyyy xx PVN yyyy	
...	...	

Table 109: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit.
...	...	
KEY		Set function key assignments.
	xx SCR yyyy	xx SCR yyyy = Key number, Single Call Ringing, DN.
	xx MCR yyyy	xx MCR yyyy = Key number, Multiple Call Ringing, DN.
	xx PVR yyyy	xx PVR yyyy = Key number, Private Line Ringing, DN.
	xx MSB	xx MSB = Key number, Make Set Busy.
...	...	

Feature operation

No specific operating procedures are required to use this feature.

Chapter 50: Malicious Call Trace

Contents

This section contains information on the following topics:

[Feature description](#) on page 355

[Operating parameters](#) on page 358

[Feature interactions](#) on page 359

[Feature packaging](#) on page 362

[Feature implementation](#) on page 362

[Feature operation](#) on page 368

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 winking) 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.

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; 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.

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 ORIGN, 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.

ACD telephones

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 110: LD 10](#) on page 363
Enable MCT on an analog (500/2500-type) telephone.
2. [Table 111: LD 11](#) on page 363
Enable MCT on a proprietary telephone.
3. [Table 112: LD 17](#) on page 364
Allow printing of the MCT record on a dedicated MCT TTY port.
4. [Table 113: LD 16](#) on page 364
Set up the recorder route.
5. [Table 114: LD 14](#) on page 364
Set up the recorder trunk.
6. [Table 115: LD 15](#) on page 365
Set up the recorder and alarm options.
7. [Table 116: LD 16](#) on page 365
Set up the alarm for external calls.
8. [Table 117: LD 57](#) on page 366
Define the MCT FFC.
9. [Table 118: LD 16](#) on page 366

Configure the call trace string.

10. [Table 119: LD 14](#) on page 367

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

11. [Table 120: LD 73](#) on page 367

Define the DTI2 flash time range.

12. [Table 121: LD 16](#) on page 368

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 110: 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.
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 111: 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.
CLS	(MCTD) MCTA	Malicious Call Trace is allowed if Class of Service is MCTA.
...		

Prompt	Response	Description
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 112: 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 113: LD 16

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

Table 114: LD 14

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RCD	Recorder 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.
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.

Table 115: 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.
...		
- 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 116: 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 117: 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 118: 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. • 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.

Prompt	Response	Description
...		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 119: 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.
XTRK	EXUT XCOT	Card type
FWTM	(NO) YES	Firmware timing for the trunk hook flash is available. This prompt is set to YES if firmware timing for trunk hook flash is supported by the card.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.

Table 120: 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.
...		

Prompt	Response	Description
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 121: 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 51: Malicious Call Trace DN/TN Print

Contents

This section contains information on the following topics:

[Feature description](#) on page 369

[Operating parameters](#) on page 369

[Feature interactions](#) on page 370

[Feature packaging](#) on page 370

[Feature implementation](#) on page 370

[Feature operation](#) on page 370

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 355 feature in this document.

Feature interactions

See the [Malicious Call Trace](#) on page 355 feature in this document.

Feature packaging

Malicious Call Trace (MCT) package 107.

Feature implementation

Table 122: 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#xxxxxxxxxxxxxxxxxx

Second Line

Field No. Field Type Contents

1. Originator<*>DNxxxxxxx+++++
2. Terminator<*>DNxxxxxxx+++++

or it could be of the following combinations of a DN and CLID number:

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

or

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

Chapter 52: Malicious Call Trace Idle

Contents

This section contains information on the following topics:

[Feature description](#) on page 373

[Operating parameters](#) on page 373

[Feature interactions](#) on page 374

[Feature packaging](#) on page 374

[Feature implementation](#) on page 375

[Feature operation](#) on page 377

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 123: LD 10](#) on page 375
Enable MCT on an Analog (500/2500-type) telephone.
2. [Table 124: LD 11](#) on page 376
Enable MCT on a proprietary telephone.
3. [Table 125: LD 15](#) on page 376
Enable the MCT signal.
4. [Table 126: LD 94](#) on page 376
Create or modify the MFC tables.
5. [Table 127: LD 16](#) on page 376
Create or modify data for each DID trunk route data block to allow or deny MFC Signaling option.
6. [Table 128: LD 14](#) on page 377
- Create or modify data for each DID trunk data block to allow or deny MFC Signaling option.

Table 123: 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 124: LD 11

Prompt	Response	Description
CLS	(MCTD) MCTA	Class of Service. Malicious Call Trace (denied) allowed. When MCTD is assigned, the MCT key is removed.
KEY	xx TRC	MCT Key number.

Table 125: 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 126: 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 127: 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 128: LD 14

Prompt	Response	Description
CLS	(DIP) DTN MFC	Class of Service. 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 53: Malicious Call Trace on Direct Inward Dialing

Contents

This section contains information on the following topics:

[Feature description](#) on page 379

[Operating parameters](#) on page 379

[Feature interactions](#) on page 380

[Feature packaging](#) on page 380

[Feature implementation](#) on page 380

[Feature operation](#) on page 381

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 outputted 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 129: LD 10](#) on page 380
- Enable MCT on an analog (500/2500-type) telephone data block.
2. [Table 130: LD 11](#) on page 381
Enable MCT on a Meridian 1 proprietary telephone data block.
3. [Table 131: LD 15](#) on page 381
Enable MCT signal
4. [Table 132: LD 94](#) on page 381
Create or modify the MFC tables.

Table 129: LD 10

Prompt	Response	Description
... CLS	MCTA	Class of Service. 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 130: 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 131: LD 15

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	Change existing data.
TYPE:	FTR	Features and options
...		
OPT	MCTA	Malicious Call Trace signal is allowed for attendants at this customer location.

Table 132: 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 54: Manual Line Service

Contents

This section contains information on the following topics:

[Feature description](#) on page 383

[Operating parameters](#) on page 383

[Feature interactions](#) on page 384

[Feature packaging](#) on page 385

[Feature implementation](#) on page 385

[Feature operation](#) on page 385

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 133: 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, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, 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 55: Manual Service Recall to Attendant

Contents

This section contains information on the following topics:

[Feature description](#) on page 387

[Operating parameters](#) on page 387

[Feature interactions](#) on page 388

[Feature packaging](#) on page 388

[Feature implementation](#) on page 388

[Feature operation](#) on page 388

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 134: 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 56: Manual Signaling (Buzz)

Contents

This section contains information on the following topics:

[Feature description](#) on page 389

[Operating parameters](#) on page 389

[Feature interactions](#) on page 390

[Feature packaging](#) on page 390

[Feature implementation](#) on page 390

[Feature operation](#) on page 391

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 135: 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
KEY	l s c u xx SIG yyy...y	Format for Large System, CS 1000E system, and MG 1000B where l = loop, s = shelf, c = card, u = unit. 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 57: Manual Trunk Service

Contents

This section contains information on the following topics:

[Feature description](#) on page 393

[Operating parameters](#) on page 393

[Feature interactions](#) on page 394

[Feature packaging](#) on page 394

[Feature implementation](#) on page 394

[Feature operation](#) on page 396

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 136: LD 16](#) on page 394
Add or change an incoming manual trunk route.
2. [Table 137: LD 14](#) on page 395
Add or change an incoming manual trunk.
3. [Table 138: LD 16](#) on page 395
Add or change an outgoing manual trunk route.
4. [Table 139: LD 14](#) on page 396
Add or change an outgoing manual trunk.

Table 136: 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
TKTP	0-511	Range for Large System, and CS 1000E system.
ICOG	TIE	Incoming manual trunks (must be TIE trunks).
ACOD	ICT	Incoming route.
	xxxx . . x	Trunk route access code.

Table 137: 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.
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.
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 138: 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.
TKTP	aaa	Outgoing trunk type, where:

Prompt	Response	Description
ICOG	OGT	aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, ISL, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT. Outgoing route.
ACOD	xx . . x	Trunk route access code.
MANO	YES	Enable manual outgoing trunk route.

Table 139: 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.
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.
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 58: Meridian 1 Attendant Console Enhancements

Contents

This section contains information on the following topics:

[Feature description](#) on page 397

[Operating parameters](#) on page 408

[Feature interactions](#) on page 412

[Feature packaging](#) on page 417

[Feature implementation](#) on page 417

[Feature operation](#) on page 419

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 398 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.

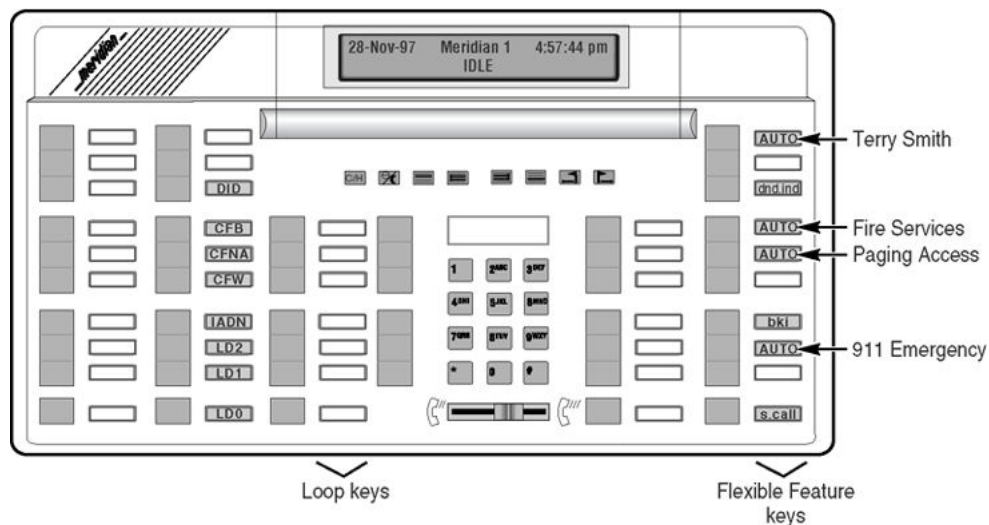


Figure 24: Attendant console with four Autoline keys configured

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 399 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 399.



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 399 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 399 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 399. 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 (Avaya 2250 Attendant Console) 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 400 illustrates an attendant console with an IADN ICI key configured on Key Strip 2. This key strip holds the ICI keys.

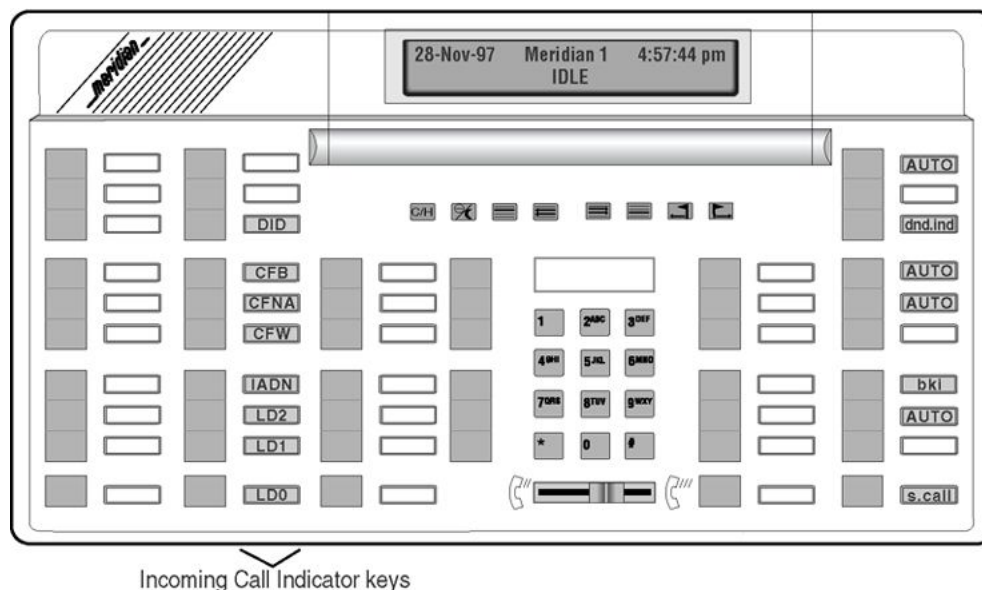


Figure 28: Attendant console with an IADN ICI key configured

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 404 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 404, 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.

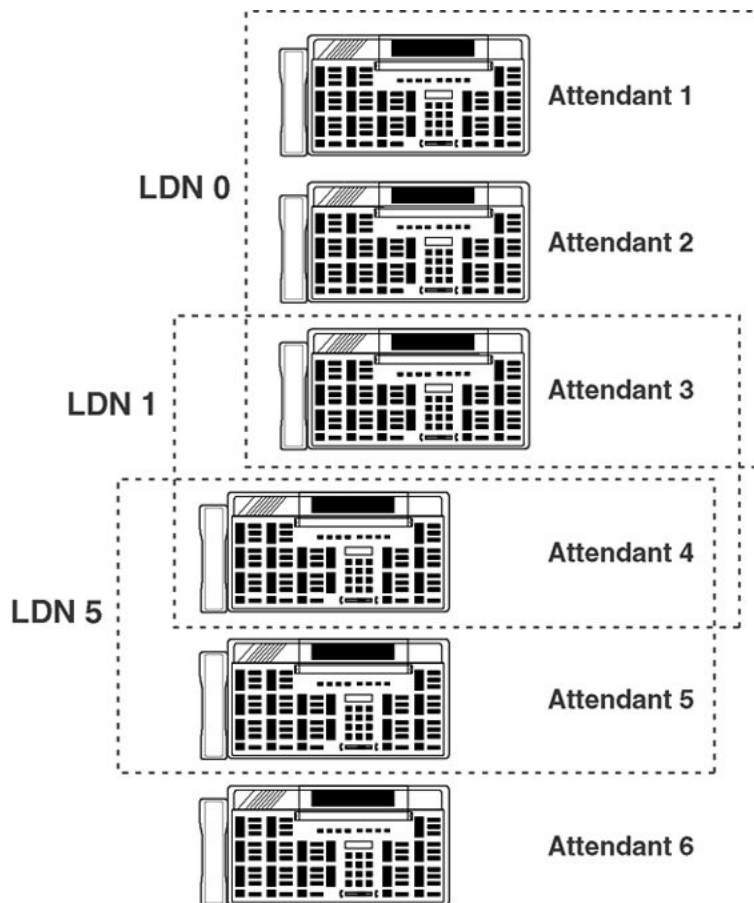


Figure 29: An example of attendant console DLDN groupings

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 405 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.

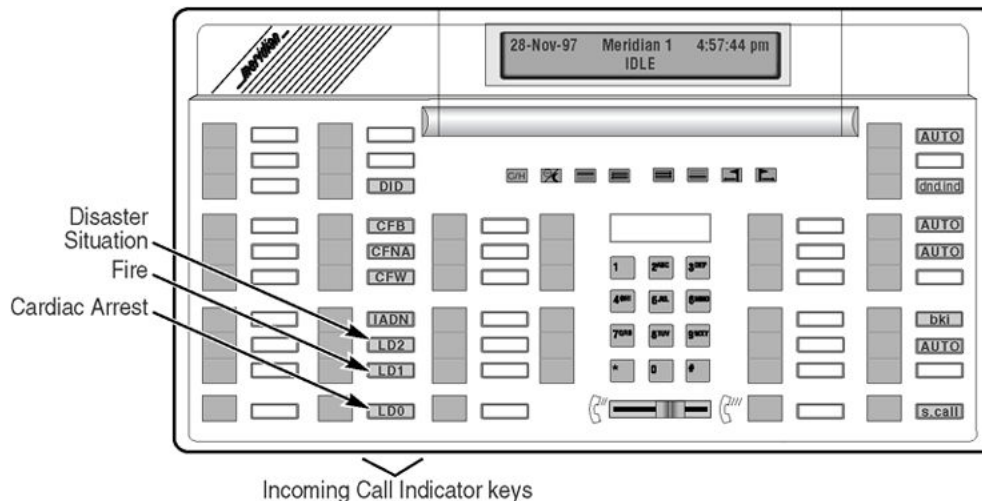


Figure 30: Attendant console with three LDN ICI keys configured

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 404, 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 404, 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 404, 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 404, 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 (Avaya 2250 Attendant Console) 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 (Avaya 2250 Attendant Console) 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 140: LD 15](#) on page 417
Configure Priority Buzzing and an Individual Attendant Directory Number (IADN) Incoming Call Indicator (ICI) key for a digital attendant console.
2. [Table 141: LD 15](#) on page 418
Configure Departmental Listed Directory Number (DLDN) and Priority Buzzing for Attendant Emergency Code calls.
3. [Table 142: LD 12](#) on page 418
Configure an Autoline DN for an attendant console.
4. [Table 143: LD 12](#) on page 419
Configure an Individual Attendant Directory Number (IADN) for a digital attendant console.

Table 140: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.

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 141: 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	Avaya 2250 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	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 142: LD 12

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

Prompt	Response	Description
TN	I s c u	Terminal number Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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 143: LD 12

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN	I s c u	Terminal number Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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 404:

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 59: Meridian 1 Initialization Prevention and Recovery

Contents

This section contains information on the following topics:

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[Feature packaging](#) on page 425

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[Feature operation](#) on page 425

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 60: Meridian 911

Contents

This section contains information on the following topics:

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Feature description

The number 911 has been adopted for the purpose of reporting emergencies and requesting emergency services. For localities with 911 systems, the number:

- is the same in all communities
- is easily remembered, even under adverse conditions
- provides direct telephone access to emergency services regardless of the time of day, or the caller 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 *Avaya 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.

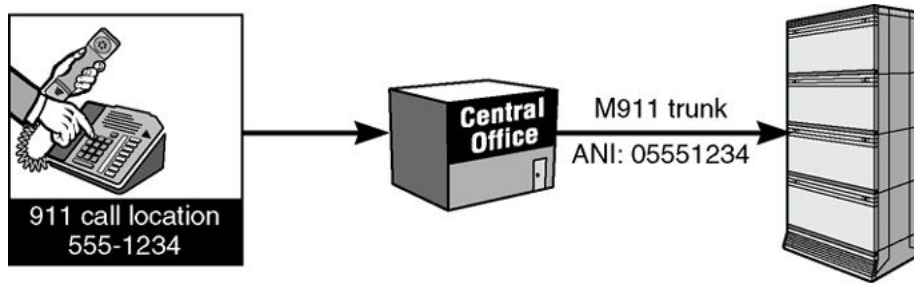


Figure 31: Routing the call, along with the ANI digits, to the system

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 *Avaya Emergency Services Access Fundamentals, NN43001-613* and *Avaya 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 (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 DN's 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 are 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 144: LD 10](#) on page 443

Configure a Terminal Number for an analog (500/2500-type) telephone with a Class of Service of CWNA (Call Waiting Notification Allowed).

2. [Table 145: LD 23](#) on page 444

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 146: LD 23](#) on page 444
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 147: LD 16](#) on page 445
Configure an M911 route.
5. [Table 148: LD 16](#) on page 445
Create a Numbering Plan or Information Digit (NPID) Table.
6. [Table 149: LD 14](#) on page 446
Configure 911 trunks.
7. [Table 150: LD 16](#) on page 446
Configure Call Detail Recording (CDR).
8. [Table 151: LD 17](#) on page 447
Configure the insertion of ANI digits into the CDR record.
9. [Table 152: LD 10](#) on page 447
Configure non-ACD sets (analog [500/2500-type] telephones).
10. [Table 153: LD 11](#) on page 447
Configure non-ACD sets (Meridian 1 proprietary telephones).
11. [Table 154: LD 11](#) on page 448
Configure Meridian 1 proprietary telephones to function as ACD sets.
12. [Table 155: LD 16](#) on page 449
Enable M911 Call Abandon.
13. [Table 156: LD 56](#) on page 449
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 144: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, 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 145: 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 146: LD 23

Prompt	Response	Description
REQ	NEW	New.
TYPE	CDN	Controlled Directory Number Data Block.

Prompt	Response	Description
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 147: 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.
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 148: 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.
TRMT	(NONE) NPA FAIL	Numbering Plan Digit or Information Digit treatment.

Prompt	Response	Description
- NPA	TEST nnn	Numbering Plan Area. Prompted only if TRMT = NPA.

Table 149: LD 14

Prompt	Response	Description
REQ	NEW	New.
TYPE	DID	Meridian 911 trunks must be DID.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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.
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 150: 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.

Prompt	Response	Description
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 151: LD 17

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

Table 152: 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.
CDEN	(DD) SD 4D	(Double), single and quadruple card density.
CUST	xx	Customer number, as defined in LD 15
DIG	xx yy	Dial Intercom Group number and Member number.
DN	nn...n	Directory Number.
...		
IAPG	2	ISDN/AP status message group.
...		
CLS	USMA	Unsolicited Status Allowed Class of Service. M911 position.

Table 153: LD 11

Prompt	Response	Description
REQ:	NEW	Add a telephone.

Prompt	Response	Description
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.
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 154: 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.
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.

Prompt	Response	Description
...		
KEY	0 ACD yyyy	Key 0; ACD; ACD Directory Number.
KEY	xx TRC	Malicious Call Trace key. The xx is the key number.

Table 155: 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.
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_ABA N	(NO) YES	Optional call abandon treatment, where: YES = abandoned call treatment for this route, and NO = no abandoned call treatment for this route.
M911_TO NE	(YES) NO	Optional call abandon tone, where: YES = tone given on answer, and NO = silence given on answer.

Table 156: 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.

Prompt	Response	Description
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 450 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.

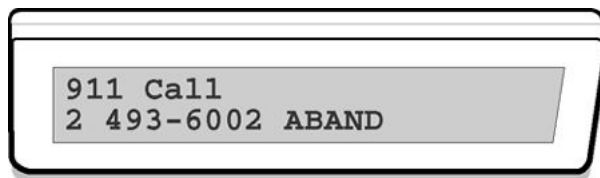


Figure 32: Display for an NPD call

[Figure 33: Display for an NPA call](#) on page 451 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.

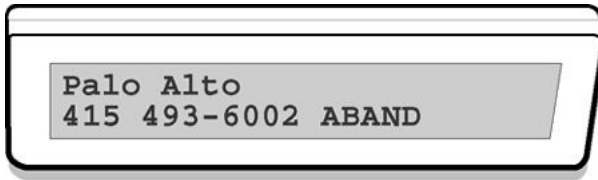


Figure 33: Display for an NPA call

Chapter 61: Message Intercept

Contents

This section contains information on the following topics:

[Feature description](#) on page 453

[Feature operation](#) on page 457

[Operating parameters](#) on page 459

[Feature packaging](#) on page 455

[Feature implementation](#) on page 455

[Feature operation](#) on page 457

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, 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.

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 157: LD 56](#) on page 456

Configure Message Intercept tones.

2. [Table 158: LD 10](#) on page 457
 - Allow Message Interception on an analog (500/2500-type) telephone.
3. [Table 159: LD 11](#) on page 457

Allow message intercept on a Meridian 1 proprietary telephone.

Table 157: 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 158: 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 159: 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 62: Message Registration

Contents

This section contains information on the following topics:

[Feature description](#) on page 459

[Operating parameters](#) on page 459

[Feature interactions](#) on page 461

[Feature packaging](#) on page 462

[Feature implementation](#) on page 462

[Feature operation](#) on page 558

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 increments 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 QPC330 trunk card must be used for the CO trunk routes receiving Reverse Battery Signals (RVB).

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 160: LD 16](#) on page 463

Activate Message Registration on routes.

2. [Table 161: LD 14](#) on page 463

Configure the polarity trunk.

3. [Table 162: LD 10](#) on page 464

Allow or deny analog (500/2500-type) telephones access to meters.

4. [Table 163: LD 11](#) on page 464

Allow or deny Meridian 1 proprietary telephones access to meters.

Table 160: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
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 161: 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.
CLS	(PIP) PSP	Polarity (insensitive) or sensitive. Use PSP for QPC295. Use PIP for QPC330, QPC331.

Table 162: LD 10

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
CLS	(MRD) MRA	MR (denied) or allowed.

Table 163: LD 11

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
CLS	aaa	Digit Display options, where: aaa = ADD, DDS, NDD.
	(MRD) MRA	MR (denied) allowed.
KEY	xx MRK	MR key, where: xx = key number.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 63: Message Waiting Indicator by Directory Number

Contents

This section contains information on the following topics:

[Feature description](#) on page 465

[Operating parameters](#) on page 471

[Feature interactions](#) on page 472

[Feature packaging](#) on page 473

[Feature implementation](#) on page 473

[Feature operation](#) on page 478

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 DN's 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 DN's 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 DN's for several individuals. [Figure 34: Multiple message waiting indications on one telephone](#) on page 467 shows a scenario where an administrative assistant monitors the DN's 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.

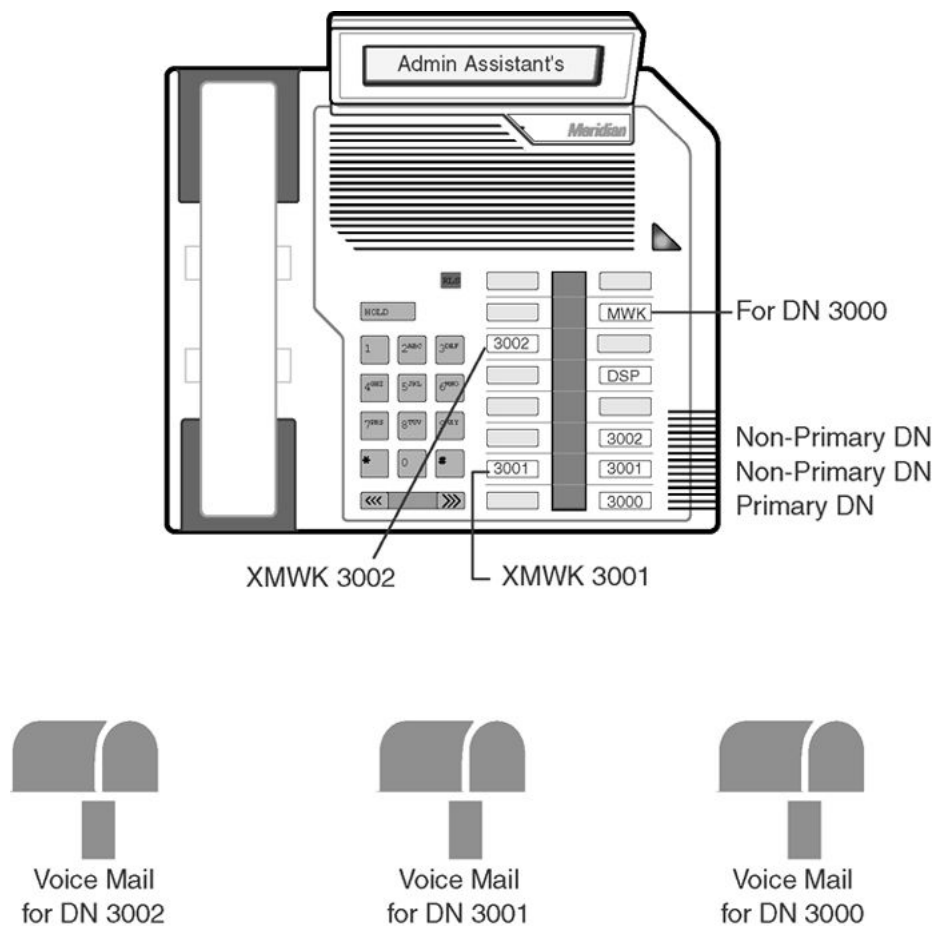


Figure 34: Multiple message waiting indications on one telephone

[Figure 35: Multiple message waiting indications for one mailbox on more than one telephone](#) on page 468 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.

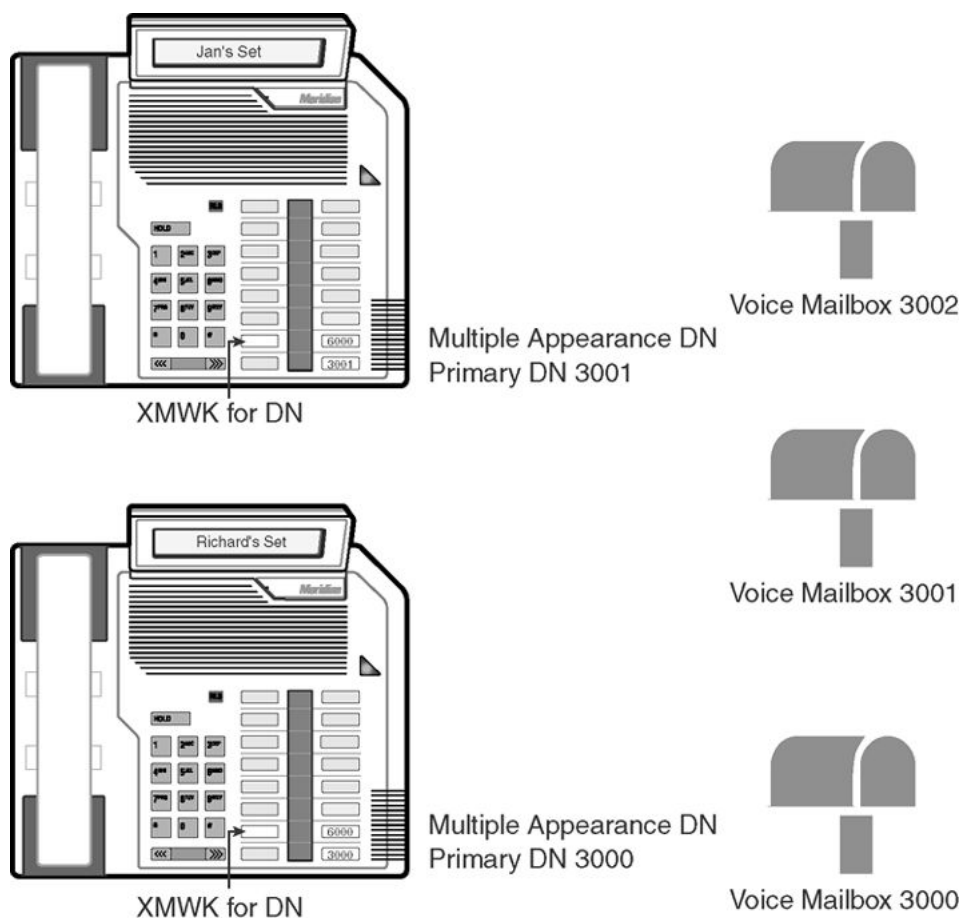


Figure 35: Multiple message waiting indications for one mailbox on more than one telephone

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 DN's 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 36: RMWK operation when a Phantom TN is call forwarded to a telephone](#) on page 470 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 37: One mailbox for multiple DNs](#) on page 471 shows DNs 5000, 5001 and 5002 with a shared mailbox.

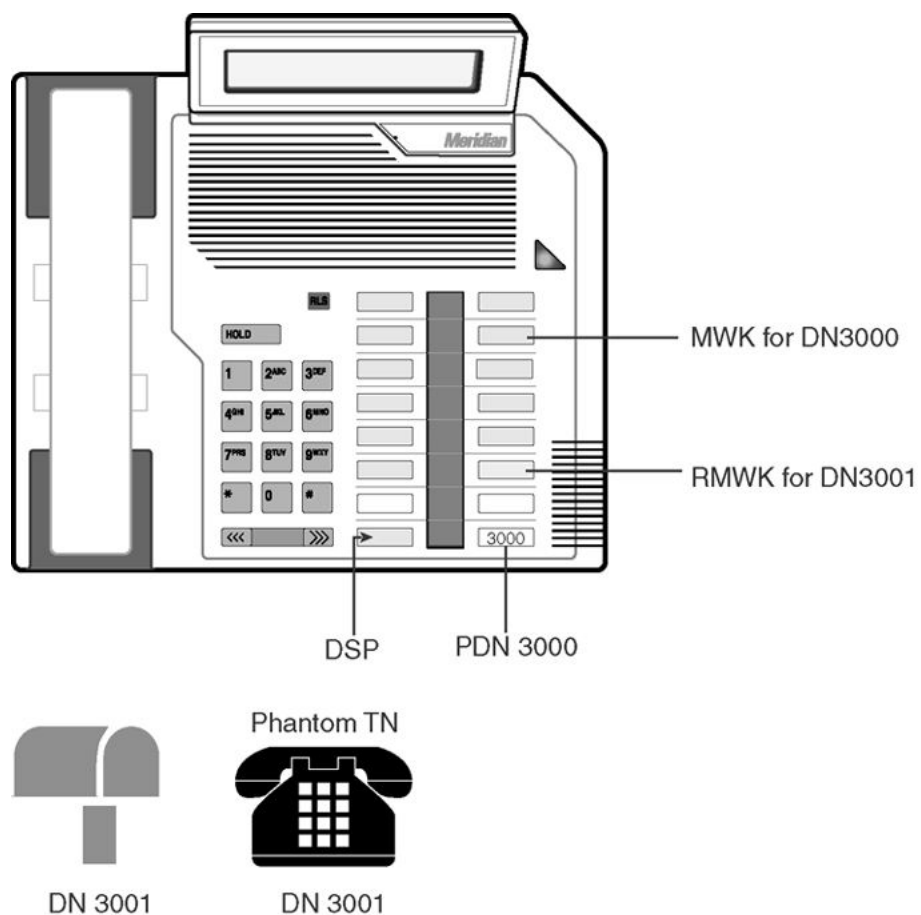


Figure 36: RMWK operation when a Phantom TN is call forwarded to a telephone

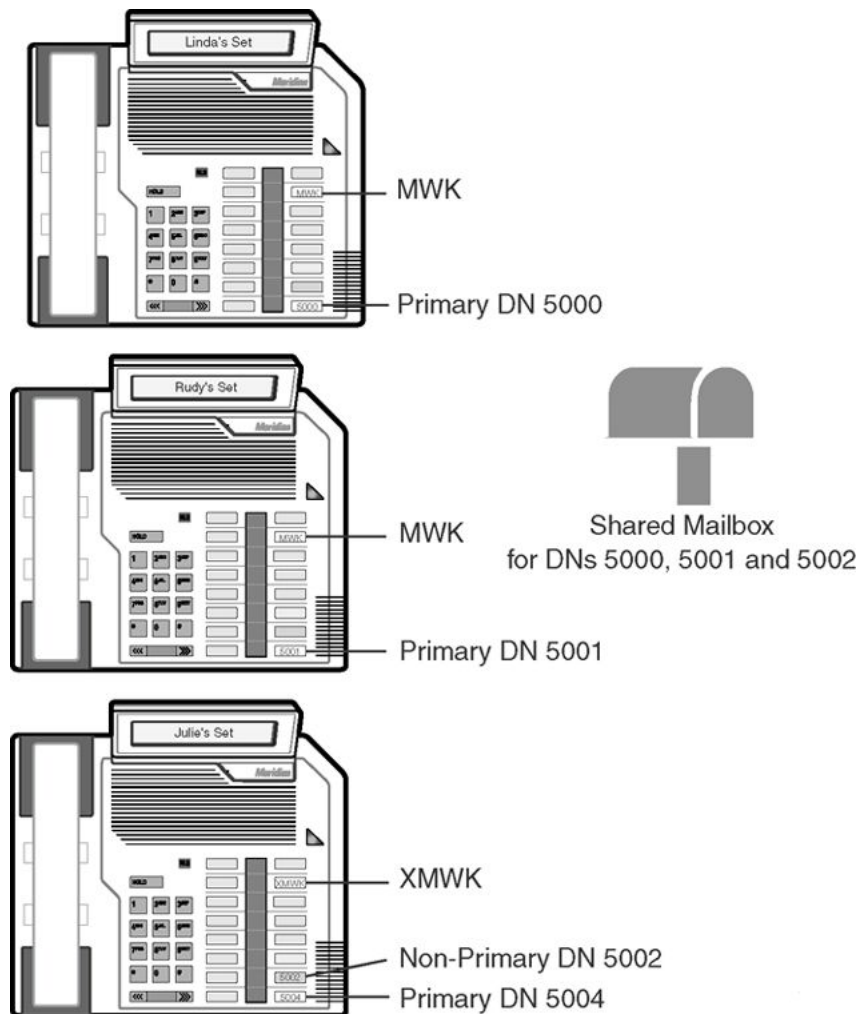


Figure 37: One mailbox for multiple DNs

Operating parameters

MWDN supports features within the same node; it does not support features on different nodes across a network.

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 164: LD 15](#) on page 474
Enable the Message Center in the Customer Data Block
2. [Table 165: LD 11](#) on page 475
Configure the class of service option for the XMWK
3. [Table 166: LD 11](#) on page 475
Configure the Remote Message Waiting Key (RMWK) to monitor remote telephones with a mailbox.
4. [Table 167: LD 10](#) on page 476
Configure new class of service options for analog (500/2500-type) telephones.
5. [Table 168: LD 11](#) on page 477
Configure the class of service options for Meridian 1 proprietary telephones.
6. [Table 169: LD 10](#) on page 477
Extend the message waiting indication function to all the analog (500/2500-type) telephones.
7. [Table 170: LD 11](#) on page 478
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 164: LD 15](#) on page 474.

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 165: LD 11](#) on page 475.

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 166: LD 11](#) on page 475.
- Configure new class of service options to enable analog (500/2500-type) telephones to be monitored remotely in LD 10. See [Table 167: LD 10](#) on page 476.

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 167: LD 10](#) on page 476.

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 169: LD 10](#) on page 477.
- 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 170: LD 11](#) on page 478.

Table 164: 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 165: LD 11](#) on page 475 to configure the class of service option for the Extended Message Waiting Key (XMWK) and its LEDs on the proprietary telephones.

Table 165: 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 166: 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	xx RMWK xxxx [yyyy]	Telephone function key assignments. Remote Message Waiting indication key Where: xx = key number xxxx = Message Center DN [yyyy] = DN to be monitored {optional}

All telephones with primary DN's to be monitored must be configured as RMMA/RMMO.

Use [Table 167: LD 10](#) on page 476 to configure a new class of service options to enable analog (500/2500-type) telephones to be monitored remotely.

Table 167: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, 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 DN's to be monitored must be configured as RMMA/RMMO.

Use [Table 168: LD 11](#) on page 477 to configure the class of service to enable Meridian 1 proprietary telephones to be monitored remotely.

Table 168: 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.
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 *Avaya Hospitality Features Fundamentals*, NN43001-553.

Use [Table 169: LD 10](#) on page 477 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 169: 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.
	MWA	Message Waiting Allowed. MWD = Message Waiting Denied.

Prompt	Response	Description
	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 DN's. For information on configuring VMBA. See *Avaya Hospitality Features Fundamentals*, NN43001-553.

Use [Table 170: LD 11](#) on page 478 to extend the message waiting indication function to all the Meridian 1 proprietary telephones on which the DN's sharing the mailbox are configured (whether the DN's are PDN or non-PDN).

Table 170: LD 11

Prompt	Response	Description
REQ:	NEW	Add new data.
	CHG	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 64: Message Waiting Lamp Maintenance

Content

The following are the topics in this section:

[Feature description](#) on page 481

[Operating parameters](#) on page 482

[Feature interactions](#) on page 482

[Feature packaging](#) on page 482

[Feature implementation](#) on page 482

[Feature operation](#) on page 483

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 171: 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 65: Message Waiting Unconditional

Contents

This section contains information on the following topics:

[Feature description](#) on page 485

[Operating parameters](#) on page 485

[Feature interactions](#) on page 486

[Feature packaging](#) on page 489

[Feature implementation](#) on page 489

[Feature operation](#) on page 489

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 DN's 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 DN's from the telephone. These DN's 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 172: 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 66: Mixed disk drive sizes with CP PII

Contents

This section contains information on the following topics:

[Feature description](#) on page 491

[Operating parameters](#) on page 491

[Feature interactions](#) on page 492

[Feature packaging](#) on page 492

[Feature implementation](#) on page 492

[Feature operation](#) on page 492

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.

Avaya Communication Server 1000M SG and Meridian 1 PBX 61C

The Avaya Communication Server 1000M (Avaya CS 1000M) SG and Meridian 1 PBX 61C systems use the install disk, which is modified to support the Mixed Disk Drive feature.

Feature packaging

No new software packages are introduced.

Feature implementation

The feature is automatically enabled during sysload. There is no manual intervention needed.

Feature operation

There are no specific procedures required to operate this feature.

Chapter 67: Mobile Extensions

Contents

This section contains information on the following topics:

[Feature description](#) on page 494

[Feature interactions](#) on page 505

[Operating parameters](#) on page 514

[Feature packaging](#) on page 516

[Feature implementation](#) on page 516

[Feature operation](#) on page 547

Conventions

This section uses the following conventions:

- Mobile Extensions and Mobile X will be used interchangeably
- an Avaya Communication Server 1000M (Avaya CS 1000M) and an Avaya 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.

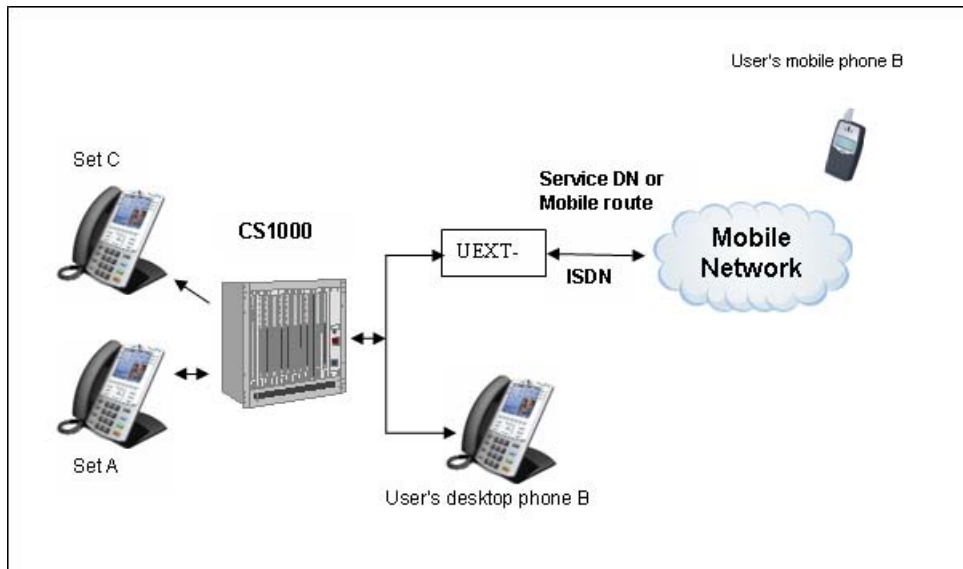


Figure 38: Mobile Extensions system overview

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 CS 1000 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.

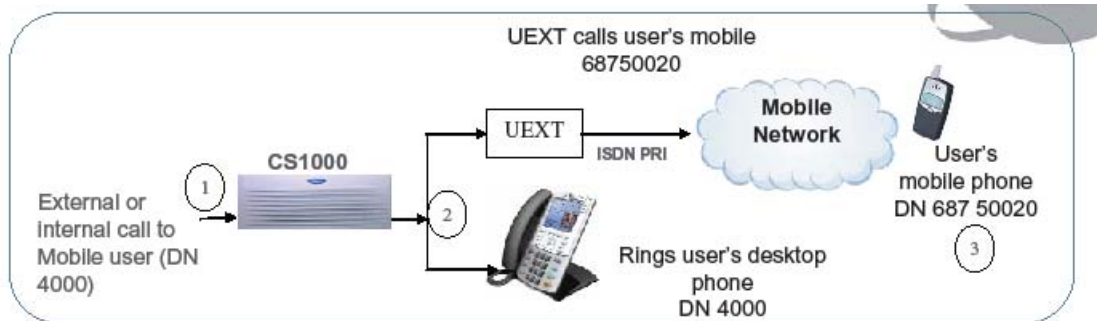


Figure 39: Call to the enterprise DN

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.

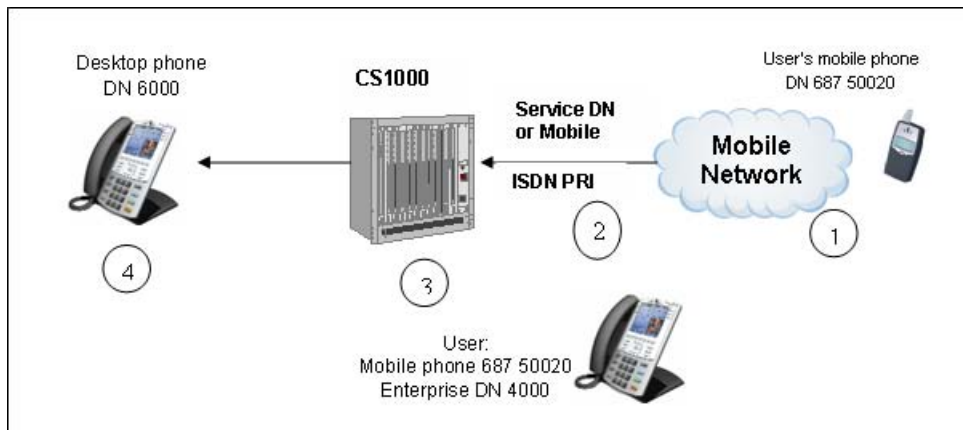


Figure 40: Call from the mobile telephone

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 173: 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
IGHA, GHDT	Group Hunt Activate and Deactivate
ICFHO	Call Forward Hunt Override
IRPAX, RPA	Dial and answer RPA (radio paging) calls

one-x mobile lite for iPhone support

The Avaya one-x mobile lite for iPhone client functions similar to other one-x mobile clients, however the iPhone API does not allow for generation of the # tones traditionally required to execute some Mobile X FCCs on a CS 1000 system. Instead of manually activating FCC features, the Avaya one-x mobile lite for iPhone has two feature buttons. The feature buttons are **extend calls to mobile** and **call forward**. Press a feature button to toggle the service on or off.

The feature buttons activate specific sequences of DTMF tones that are sent from the iPhone client to the CS 1000. The feature button DTMF tones contain an End of Dialing (EOD) sequence that you can configure the CS 1000 to detect.

The default CS 1000 EOD sequence is a # tone. To support the one-x mobile lite for iPhone client, you must configure the CS 1000 to remove dependency on # tone for the following Mobile X FCCs:

1. PCAA FCC = enable PCA (example, turn on sim ring)
2. PCAD FCC = disable PCA (example, turn off sim ring)
3. CFWA FCC = activate call forward for all calls
4. CFWD FCC = disable call forward for all calls

For information on how to configure the CS 1000 to support the one-x mobile lite for iPhone, see [Configure one-x mobile lite for iPhone](#) on page 525

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 516).

The MFAC is not valid in normal dialing sequences and therefore does not have to be valid within the customer's dialing 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 CS 1000 will be immediately answered by cellular VM in case mobile phone is turned off or out of range. The Cellular VM avoidance feature allows mobileX user to keep cellular VM configured together with CS 1000 Call Pilot VM. With proper configuration on CS 1000, cellular VM is ignored for all calls to mobileX user through CS 1000 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 CS 1000. It is administrator's responsibility 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 feature 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 CS 1000. 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.

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. In case 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 CS 1000. 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 *Avaya 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 *Avaya 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 dialling 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 dialling 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, you 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 CP PIV and CP PM.

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.

Communication Server 1000 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 CS 1000 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

Task	Source based routing Implementation		Mobile Service Access DN Implementation	
	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)
1.	Table 174: LD 15—Enable PCA at the Customer Level on page 519		Table 174: LD 15—Enable PCA at the Customer Level on page 519	

Task	Source based routing Implementation		Mobile Service Access DN Implementation	
	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)
2.	Table 175: LD 15—Configure MFAC for Call in progress on page 519		Table 175: LD 15—Configure MFAC for Call in progress on page 519	
3.	Table 176: LD 15—Configure ringing cycles for Call Forward No Answer on page 519		Table 176: LD 15—Configure ringing cycles for Call Forward No Answer on page 519	
4.	Table 177: LD 22—Check equipped status of the Mobile X package on page 520		Table 177: LD 22—Check equipped status of the Mobile X package on page 520	
5.	Table 178: LD 16—Configure a Dedicated Mobile Route on page 520	Configuring a new Mobile Route on page 528	N/A	N/A
6.	N/A	N/A	Table 179: LD 24—Configure a Mobile Service DN with Security Code on page 521	Adding a new Mobile Service Directory Number on page 529, Editing an existing Mobile Service Directory Number on page 530, Deleting Mobile Service Directory Number on page 531
7.	Table 180: LD 20—Print MOBX data on page 521		Table 180: LD 20—Print MOBX data on page 521	
8.	Table 181: LD 20—Print UEXT TN block on page 522		Table 181: LD 20—Print UEXT TN block on page 522	

Task	Source based routing Implementation		Mobile Service Access DN Implementation	
	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)	Using CS 1000 Overlays (CLI)	Using CS 1000 Element Manager (GUI)
9.	Table 182: LD 11—Configure Mobile X users on page 522	Basic Client Configuration (BCC) on page 541, Configuring Features and Keys on page 542, Copying Class of Services from Desktop telephone to UEXT-MOBX on page 544, Configuring Features and Keys on page 545	Table 182: LD 11—Configure Mobile X users on page 522	Basic Client Configuration (BCC) on page 541, Configuring Features and Keys on page 542, Figure 61: Copy Class of Service from desktop telephone on page 545, Configuring Features and Keys on page 545
10.	Table 183: LD 11—Configure a Handoff key for a Mobile X desktop phone on page 523		Table 183: LD 11—Configure a Handoff key for a Mobile X desktop phone on page 523	
11.	Table 184: LD 57—Configure Flexible Feature Codes for Mobile Call In Progress on page 523	Flexible Feature Code Entries on page 533, Searching for Flexible Feature Codes on page 533, Searching for Flexible Feature Codes by Value on page 534, Searching for Flexible Feature Codes (Advanced) on page 535, Configuring Flexible Feature Codes (Basic) on page 535,	Table 184: LD 57—Configure Flexible Feature Codes for Mobile Call In Progress on page 523	Flexible Feature Code Entries on page 533, Searching for Flexible Feature Codes on page 533, Searching for Flexible Feature Codes by Value on page 534, Searching for Flexible Feature Codes (Advanced) on page 535, Configuring Flexible Feature Codes (Basic) on page 535,

	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)
		Configuring Flexible Feature Codes (Advanced) on page 535, Editable FFC codes on page 536, Editing Group Call Code on page 537		Configuring Flexible Feature Codes (Advanced) on page 535, Editable FFC codes on page 536, Editing Group Call Code on page 537

LD tables

The following tables provide Mobile Extensions implementation prompts and responses.

Table 174: 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 175: 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 176: LD 15—Configure ringing cycles for Call Forward No Answer

Prompt	Response	Description
REQ:	CHG	Change.

Prompt	Response	Description
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 177: LD 22—Check equipped status of the Mobile X package

Prompt	Response	Description
REQ	PRT	Print.
TYPE	PKG 412	List the equipped packages.

Table 178: 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.

Prompt	Response	Description
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 179: 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 180: LD 20—Print MOB data

Prompt	Response	Description
REQ	PRT	Print.
TYPE	MOBX	Print the Universal Extension data for a MOB 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 181: 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 *Avaya Software Input Output Administration* (NN43001-611), *Avaya Software Input Output Reference –Maintenance* (NN43001-711), and *Avaya Electronic Switched Network Reference - Signaling and Transmission* (NN43001-280).

Table 182: 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 <i>Avaya Flexible Feature Codes in the Features and Services Fundamentals</i> (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

Prompt	Response	Description
		to the Mobile Network. (For example: TSC (Trunk Steering Code) <zzzz> is the mobile telephone DN.

Table 183: 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 *Avaya Personal Call Assistant in the Features and Services Fundamentals* (NN43001-106) Book 5 of 6.

Table 184: 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.

Prompt	Response	Description
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 185: 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.
DSEL		

Table 186: 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 187: 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.

Prompt	Response	Description
...		
CLS	CCMA (or CCMD)	Cell CLID Mode Allowed (or Denied).

Table 188: 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 189: 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.

Configuration of one-x mobile lite for iPhone

The iPhone API does not allow for the generation of # tones. Due to this restriction, you must specifically configure the CS 1000 to support the call forward and sim ring feature buttons of

the Avaya one-x mobile lite for iPhone client. The following is a configuration example to support the iPhone client.

Example

LD 15: FCC_DATA for MOBX_FCCS enter YES

Input is accepted.

STRL is prompted.

STRL 7 (Enter a DTMF string length used to replace the # key. It can be 1–7 and must be part of the dial plan.

STRG is prompted.

STRG 1234567 (Enter the DTMF sequence; string must match the STRL length, in this example, 7 digits)

You must ensure that PCAA, PCAD, CFWA, and CFWD FCCs are configured in LD 57 as DTMF strings without * and # keys. The four FCCs must be configured for DTMF strings using only digits 0–9 to support the one-x mobile lite for iPhone client.

You must choose a valid EOD (MOBX_FCCS) string to avoid CFW failure. You must avoid the use of the following MOBX FCC string types:

1. 9999999 or 0000000 or any order same number sequence. CFW failure occurs for all CFW numbers ending in 9 or 0.
2. 1234123 or any repetitive string sequence. CFW failure occurs for CFW number xxx1234 because xxx1234+1234123 (EOD) cuts the CFW number to xxx.

Note:

Each FCC on a CS 1000 system can have multiple values configured. If you use multiple values, you can keep FCC values containing * and # keys to support non-iPhone Mobile X users.

Feature limitation

If the STRG sequence is part of a CFW number (a number where the Mobile X phone is forwarded to), it will be recognized as an EOD sequence and CFW will not function. In this case, the user can perform a CFWA manually using the # key as EOD from the iPhone dialpad instead of from the one-x mobile lite client

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 516.
- 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 *Avaya 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 a configured customer.
3. Choose 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 select **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 appears and it is not selected by default.
6. If the **Mobile Extension Route** prompt is checked, the sub prompt **Screen Indicator** appears and it is not selected by default.
7. Type the values for the mandatory fields, and click **Submit**.
8. The Routes and Trunks page is loaded with the new route entry (see [Figure 41: Routes and Trunks](#) on page 528).

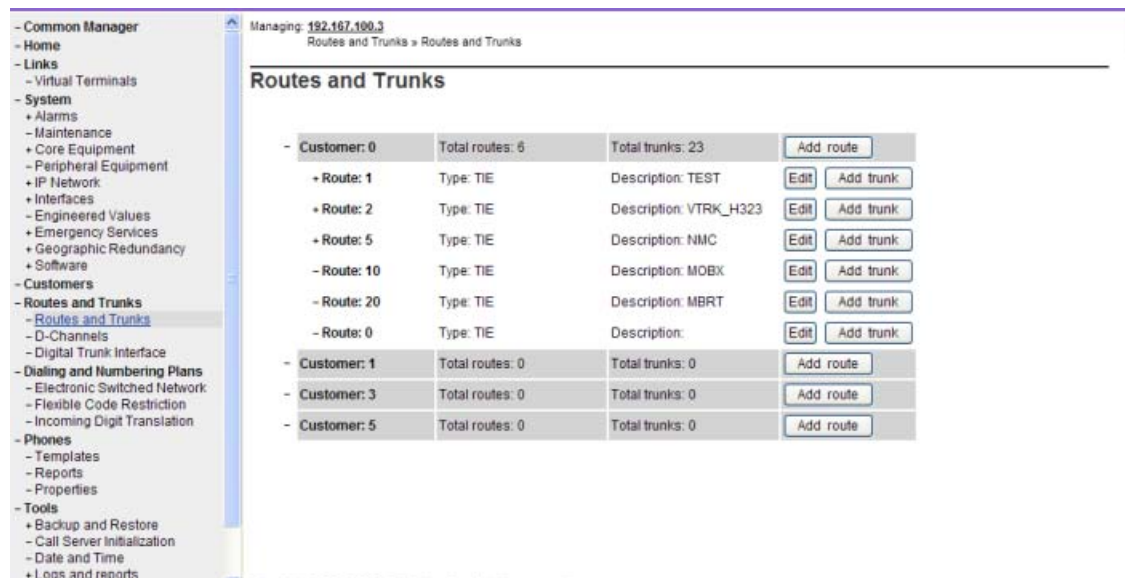


Figure 41: Routes and Trunks

Mobile Service Directory Numbers page

1. From the Navigation tree, click **Customers**.
2. Click # (customer number) hyperlink in the data grid.

On the **Edit** page, a new hyperlink Mobile Service Directory Numbers link appears.

3. Click a hyperlink.

A new Mobile Service Directory Numbers page appears, as shown in the following figure.

This page displays the list of configured Mobile Service Directory Numbers in a datagrid.



Figure 42: Mobile Service Directory Numbers link

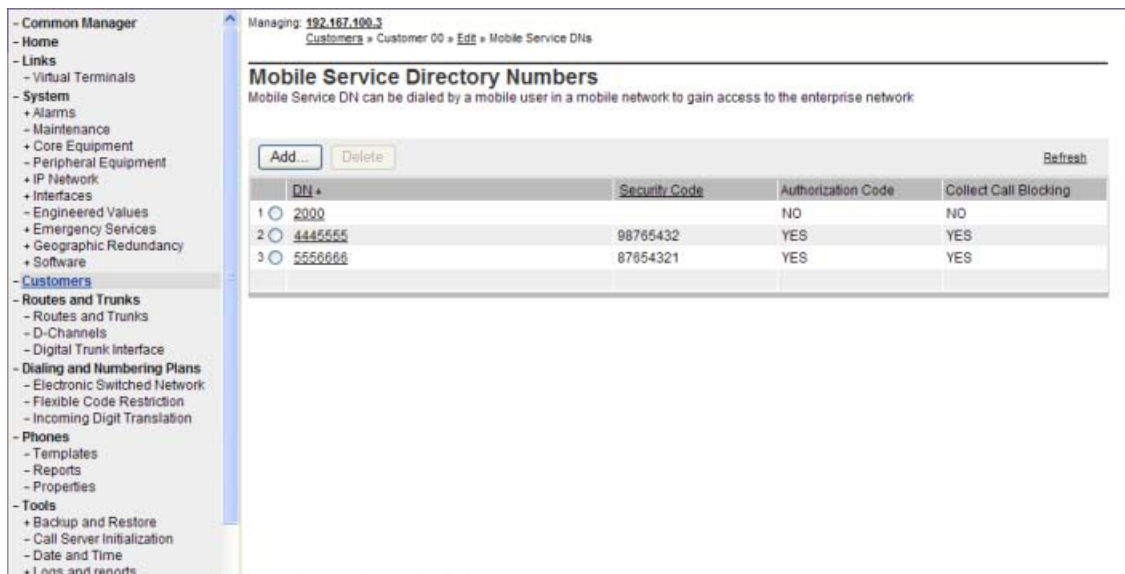


Figure 43: Mobile Service Directory Numbers page

Adding a new Mobile Service Directory Number

1. From the navigation tree, click **Customers**.
2. Click a # (customer number) hyperlink in the data grid.
3. On the Edit page, click Mobile Service Directory Numbers hyperlink to navigate to the Mobile Service Directory Numbers page.
4. Click **Add**.

The Add Mobile Service Directory Number page appears, as shown in [Figure 44: Add Mobile Service Directory Numbers page](#) on page 530.

5. Type the **Mobile service directory number** (up to 7 digits).
6. Type a **Security code** (up to 8 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 appears.

9. Click **Save**.

The Mobile Service Directory Numbers page appears.

Figure 44: Add Mobile Service Directory Numbers page

Editing an existing Mobile Service Directory Number

1. From the navigation tree, click **Customers**.
2. Click a # (customer number) hyperlink in the data grid.
3. On the Edit page, click the Mobile Service Directory Numbers hyperlink to navigate to the Mobile Service Directory Numbers page.
4. Click a # (Mobile Service Directory Number) hyperlink in the data grid.

The Edit Mobile Service Directory Number page appears, as shown in [Figure 45: Edit Mobile Service Directory Numbers page](#) on page 531 .

5. Click **Save**.

The Mobile Service Directory Numbers page appears.

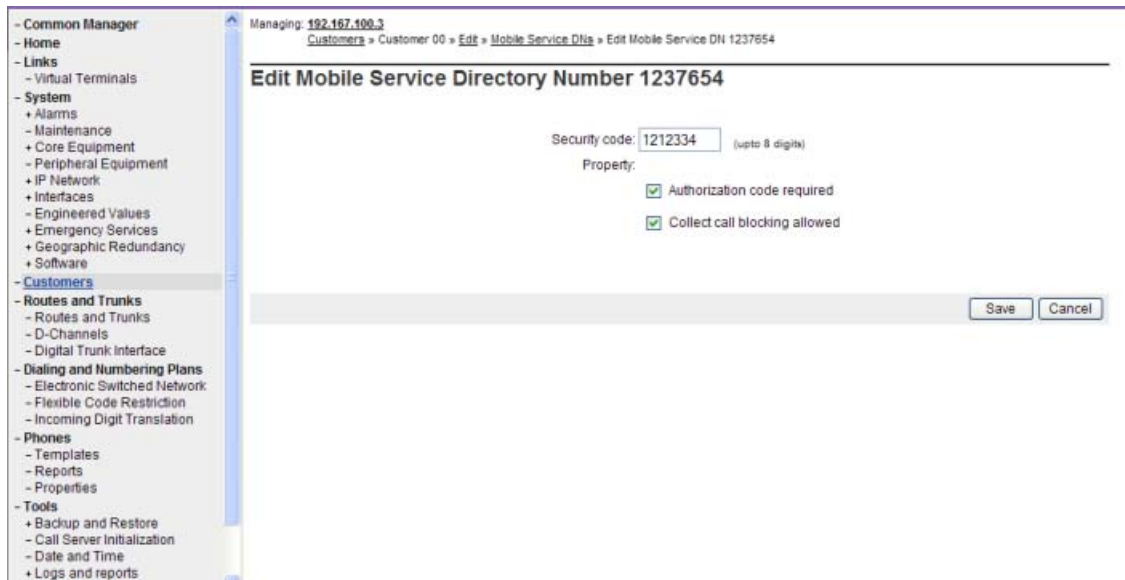


Figure 45: Edit Mobile Service Directory Numbers page

Deleting Mobile Service Directory Number

1. From the navigation tree, click **Customers**.
2. Click # (customer number) hyperlink in the data grid.
3. On the **Edit** page, click the Mobile Service Directory hyperlink to navigate to Mobile Service Directory Numbers page.
4. Select an item, and click **Delete**.

The selected entry is deleted and the page is refreshed.

Password protection for Mobile Service DN configurations

The Password window appears if the Secure Data Password is configured in the password block for the Customer in LD 15.

After the password is typed, 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. From the navigation tree, click **Customers**.
2. Click # (customer number) hyperlink in the data grid.
3. On 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 appears.

5. Type a **Mobile service directory number** (up to 7 digits).
6. Type a **Security code** (up to 8 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 46: Password window](#) on page 532.
9. Type the Secure Data Password, and click **Save**. The window closes.
10. 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.

The image shows a 'Password' window. It has a title bar with the word 'Password'. Below the title bar, there is a label 'Secure Data Password:' followed by a text input field. The input field contains five asterisks '*****' and a small asterisk '*' to its right, indicating a required field. At the bottom of the window, there are two buttons: 'Save' and 'Cancel'.

Figure 46: Password window

Deleting Mobile Service Directory Number

1. From the navigation tree, click **Customers**.
2. Click # (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 one of the radio buttons, and click **Delete**.

The Password window is displayed.

5. Enter the Secure Data Password, and click **Save**.

The Password window closes. 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

Use the Flexible Feature Codes Web page 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. From the navigation tree, click **Customers**.
2. Choose a customer number from the list.
3. Click the Flexible Feature Codes link.

Note:

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. From the navigation tree, click **Customers**.
2. From the **Flexible Feature Codes** page, choose a customer number.

The Flexible Feature Code Entries page appears.

3. In the **Criteria** section, type a code type or click **Lookup** or type a Value and click **Search**. New flexible feature codes may also be added as indicated in [Figure 47: Flexible Feature Code Entries page](#) on page 533 .

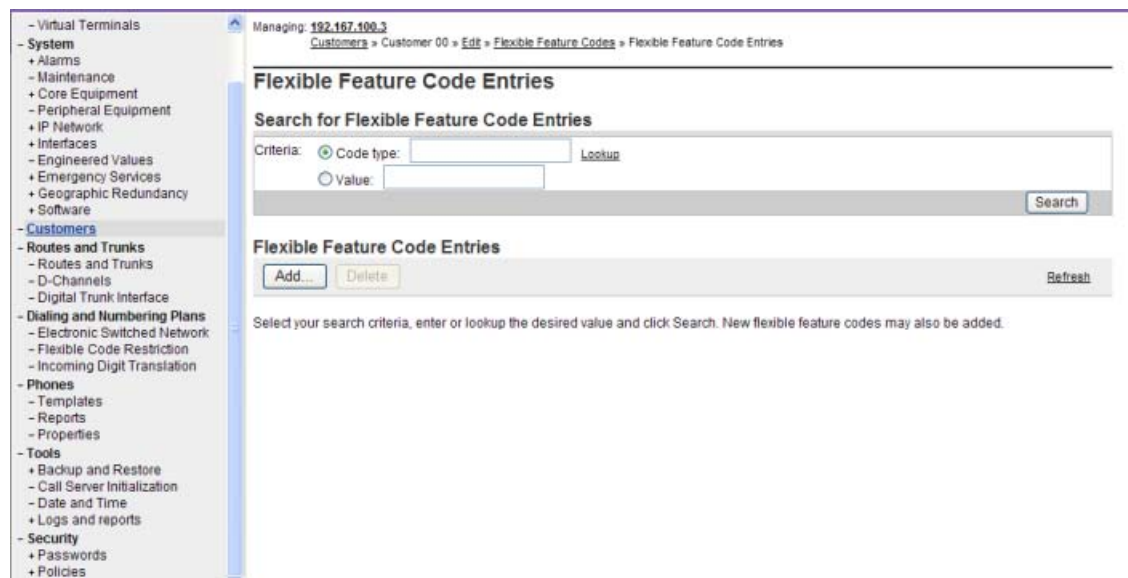


Figure 47: Flexible Feature Code Entries page

Searching for Flexible Feature Codes

1. From the navigation tree, click **Customers**.
2. Choose a customer number from the list.
3. Click the Flexible Feature Codes link.
4. Click **Lookup** in the Search block.

The Flexible Feature Code Lookup window opens. See [Figure 48: Flexible Feature Code Lookup](#) on page 534.

5. Select the check box for the required feature,s and click **Assign**.
6. The selected feature prompt names are displayed in the **Code type** input box.

- 7. Select the **Code type** radio button.
- 8. Click **Search** to display the codes for the selected feature in the datagrid. See [Figure 48: Flexible Feature Code Lookup](#) on page 534.

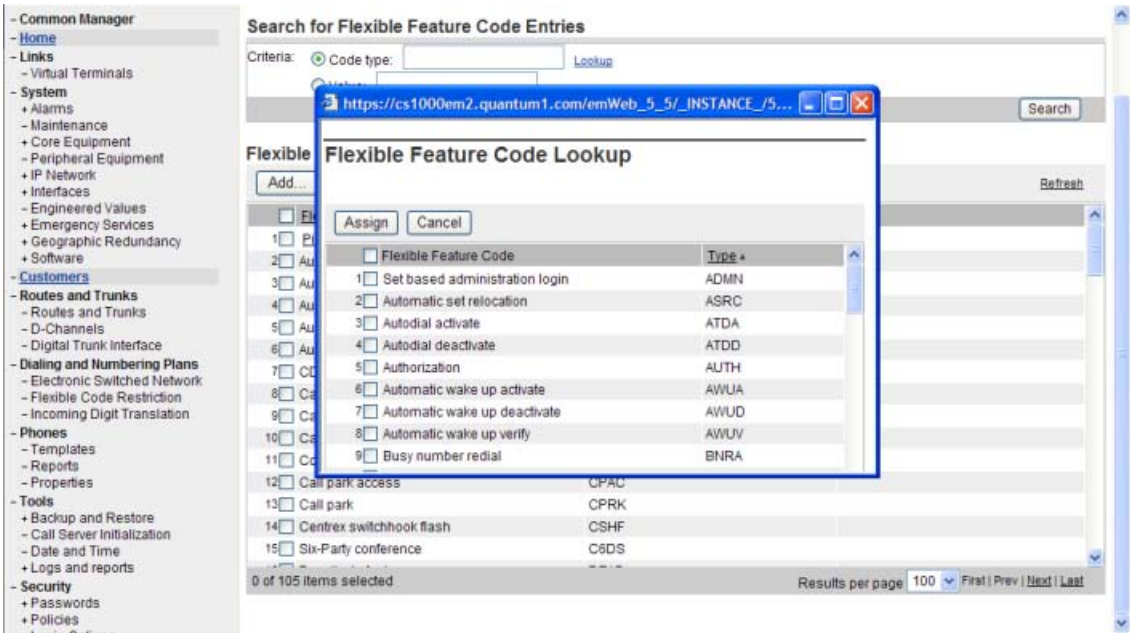


Figure 48: Flexible Feature Code Lookup

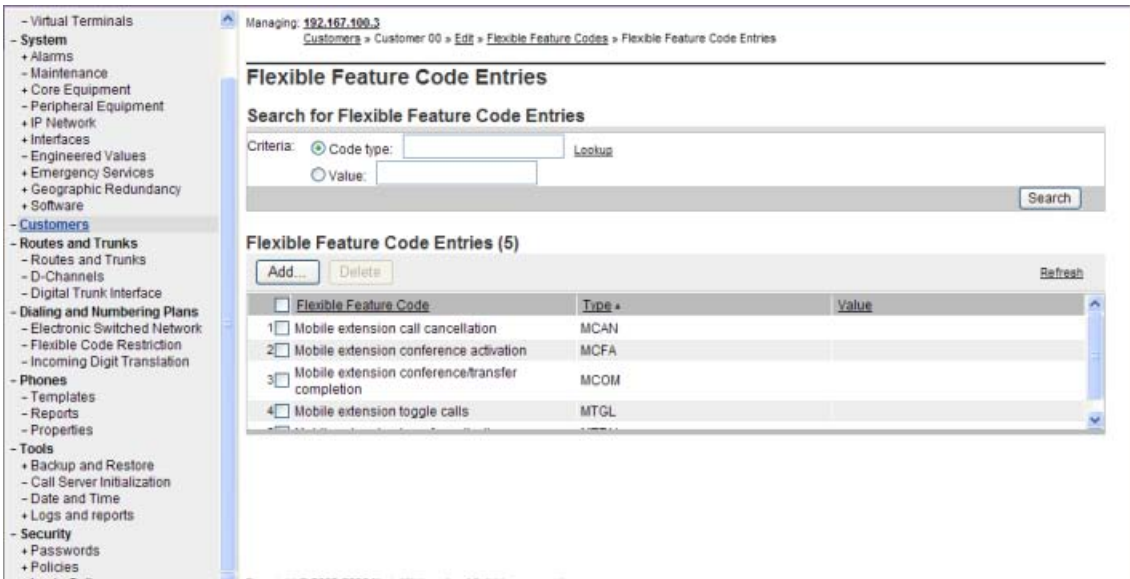


Figure 49: Flexible Feature Code Entries page after Search Operation

Searching for Flexible Feature Codes by Value

Search 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)

Search for Flexible Feature Codes using Advanced Search.

1. From the navigation tree, click **Customers**.
2. Choose a customer number from the list.
3. Click the Flexible Feature Codes link.
4. Enter the wildcard character * before or after the search text in the **Code type** input box.
5. Select the **Code type** radio button.
6. Click **Search**. All the flexible feature code types with the configured values that match the given search text are listed in the datagrid.

Note:

Flexible Feature Code configuration: Element Manager does not have the capacity to configure more than 3 000 Flexible Feature Codes (for example, 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. From the navigation tree, click **Customers**.
2. From the **Flexible Feature Codes** page, choose a customer number.
The Flexible Feature Code Entries page appears
3. Select the box **Provide confirmation tone**.
4. To complete the configuration, click **Save**.

OR

Click **Cancel** to cancel the action.

The Edit Web page displays.

Configuring Flexible Feature Codes (Advanced)

1. From the navigation tree, click **Customers**.
2. From the **Flexible Feature Codes** page, choose a customer number.
The Flexible Feature Code Entries page appears
3. Click **Add**.
The Add Flexible Feature Code Web page opens.

- Click the **Lookup** adjacent to the **Flexible Feature Code type** input box.

The Flexible Feature Code Lookup window opens. See [Figure 50: Add Flexible Feature Code](#) on page 536. The Lookup lists all the Flexible Feature Code types and their corresponding descriptions.

- Select a code type to appear in the **Flexible Feature Code type** input box or type the FFC code type.
- Enter a FFC in the range 0-##### in **Value** input box.
- Click **Save**.



Figure 50: Add Flexible Feature Code

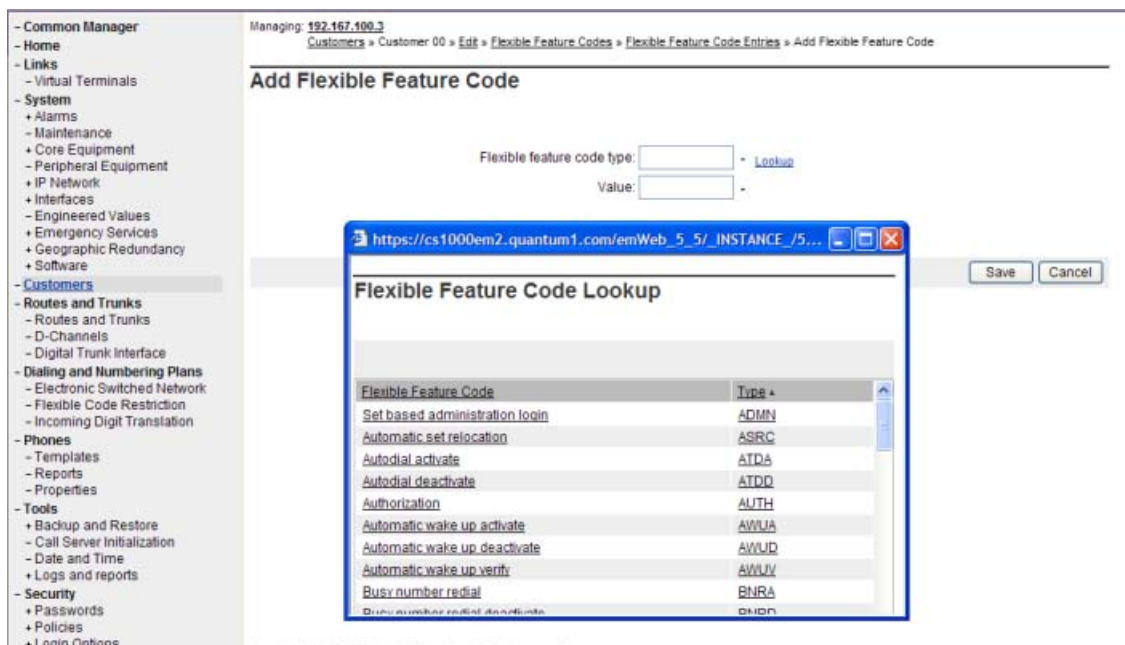


Figure 51: Add Flexible Feature Code Lookup

Editable FFC codes

- Navigate to **Flexible Feature Code Entries** page.
- Navigate to Group call code, Pilot DN code and 1xx special services.

Configuring Group Call Code

1. From the navigation tree, navigate to **Customers > Select a customer number > Flexible Feature Codes > Flexible Feature Code Entries > Add Flexible Feature Code**.
2. Click 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 completed as indicated in [Figure 52: Add Group Call code](#) on page 537.
4. Click **Save**. The Flexible Feature Code is configured and the Flexible Feature Code Entries page displays with the new entry.

OR

Click **Cancel**.

The Flexible Feature Code Entries page displays.

Figure 52: Add Group Call code

Editing Group Call Code

1. Navigate to **Flexible Feature Code Entries** page.
2. Select a configured # (GRPF).

The Edit Group Call Code xxxxxxxx page displays, as shown in [Figure 53: Edit Group Call code xxxx](#) on page 538.

3. Edit the value in **Group call list number** field.
4. Click **Save**.

The Flexible Feature Code is configured and the Flexible Feature Code Entries page displays with the edited entry.

OR

Click **Cancel**.



Figure 53: Edit Group Call code xxxx

Configuring 1xx special features code

1. From the navigation tree, navigate to **Customers > Select a customer number > Flexible Feature Codes > Flexible Feature Code Entries > Add Flexible Feature Code**.
2. Select the **Lookup** hyperlink and select ITXX from the lookup table.
3. Complete the values for sub prompts **CO route number for the '1xx' service** and **Value**, as shown in [Figure 54: Add 1xx special services code](#) on page 538.
4. Click **Save**. The Flexible Feature Code is configured and the Flexible Feature Code Entries page displays with the new entry.

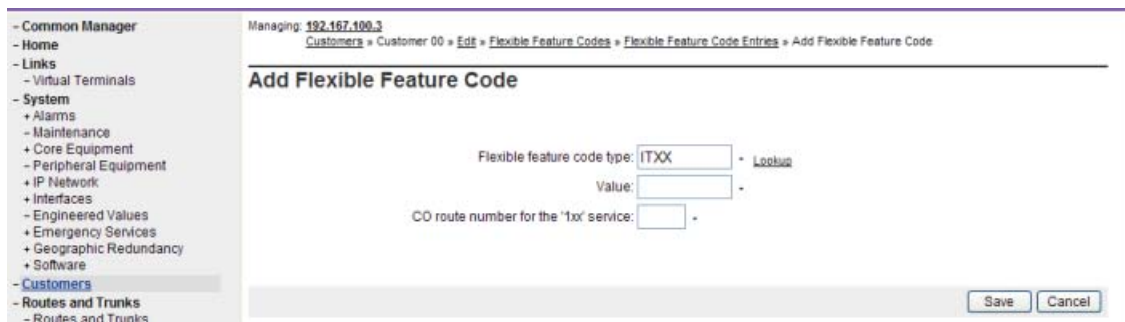


Figure 54: Add 1xx special services code

Editing 1xx special features code

1. Navigate to **Flexible Feature Code Entries** page.
2. Select a configured ITXX hyperlink.
3. The Edit For '1xx' Special Services xxxxxxxx for ITXX displays, as shown in [Figure 55: Edit for 1xx special services code page](#) on page 539 . Edit the value.
4. Click **Save**. The Flexible Feature Code is configured and the Flexible Feature Code Entries page displays with the edited entry.

Figure 55: 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. Click **Lookup** and select **PLDN** from the lookup table.
The **List number** and **Use** fields appear, as shown in [Figure 56: Add Pilot DN code](#) on page 539.
3. In the **List number** field, enter a value from 0–254.
4. In the **Use** field, choose **Initiate Group Hunting**.
5. Enter a value in the **Limit to calls queued against pilot DN** field.
6. Enter a value in the **Hunting type** field.
7. Select the **Property** check box if you want to enable **Call forward all calls have priority over group hunting**.
8. Click **Save**.

Figure 56: 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 the **Lookup** hyperlink and select **PLDN** from the lookup table.
The **List number** and **Use** fields appear, as shown in [Figure 56: Add Pilot DN code](#) on page 539.
3. In the **List number** field, enter a value from 0–254.
4. In the **Use** field, choose **SCL/SSC list user** or **SCL/SSC list controller**.
5. Enter a value in the **Limit to calls queued against pilot DN** field.
6. Enter a value in the **Hunting type** field.
7. Select the **Property** check box if you want to enable **Call forward all calls have priority over group hunting**.
8. Click **Save**. The Flexible Feature Code is configured and the Flexible Feature Code Entries page displays with the new entry.

Editing Pilot Directory Number code – Use as Initiate Group Hunting

1. Navigate to **Add Flexible Feature Code** page.
2. Select a PLDN hyperlink.
The Edit Pilot DNcode xxxxxxxx appears.
3. Edit the List number value.
- 4.
5. Click **Save**.



Figure 57: 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. Select a PLDN hyperlink.
3. The Edit Pilot DN xxxxxxxx displays, as shown in [Figure 58: Edit Pilot DN code page - Use as SCL/SSC List Controller](#) on page 541.

4. Edit the List number value.
5. Click **Save**.

Figure 58: 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**, **Mobile extension call cancellation**, **Mobile Extension Conference/Transfer Completion**, **Mobile extension toggle calls**, or **Mobile Extension Transfer Activation** from the **lookup** table.
3. Enter a value in the **Value** box.
4. Click **Save**.

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, or Mobile Extension Transfer code.

Error Scenario for Mobile Flexible Feature Codes

1. Navigate to **Add Flexible Feature Code** page.
2. Select a 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**.

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 Avaya 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 Mandatory 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 for UEXT-MOBX configuration

Refer to the following procedures to configure BCC.

Configuring Features and Keys

1. Log on to UCM.
2. Select the Call Server element to launch Element Manager.
3. From the navigation pane, click **Phones**.

The Search for Phones page appears.

Note:

The telephone and call server database must be kept synchronized at all times. Customer information must be available for a telephone to be configured (for example, superloop, customer).

4. From the Search for Phones page, navigate to the **Phones** heading and click **Add**.

The New Phones page opens.

5. Select **UEXT-MOBX-Universal Extension MOBX** from the telephone **Type** list (see [Figure 59: UEXT set configuration](#) on page 543)

- Complete 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 shows 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.
- Navigate to the **Features** section. In the **UXID** field, type the **Universal Extension ID**.
 - 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.
 - Complete the required details for Key 1 and Key 2, and click **Validate** to perform validation of general properties, features and keys.
 - Click **Finish**.

The screenshot shows the 'New Phones' configuration window. On the left, there are fields for 'Number of phones' (set to 1), 'Customer' (set to 0), 'Type' (radio buttons for 'Phone Type', 'Feature Pack', and 'Copy from Template'), and 'Options' (checkboxes for 'Default value for ZONE', 'Automatically assign DN', and 'Automatically assign TN'). A dropdown menu is open, displaying a list of phone models and features. The option 'UEXT-MOBX-Universal Extension MOBX' is highlighted. At the bottom right, there are 'Preview' and 'Cancel' buttons. A legend at the bottom left indicates '* Required Value'.

Figure 59: 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 Key Type from the list.
3. Type a DN value, and click **telephone handset image** as shown in [Figure 60: Key configuration for UEXT sets](#) on page 545.
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 61: Copy Class of Service from desktop telephone](#) on page 545.
5. Select a DN and click **Assign** . The values of HUNT, FDN, CLS, NCOS & TGAR class of service values are copied from the telephone selected to the new telephone being configured.

Keys

Key No.	Key Type	Key Value
0	SCR - Single Call Ringing	DN: 2024 ☐ Multiple Appearance Redirection Prime(MARP) CLID Entry (Numeric or D) ANIE Entry
1	HOT_P - Hotline	Target DN Length 5 Target DN 122
2	NUL - Unassigned	
3	NUL - Unassigned	
4	NUL - Unassigned	
5	NUL - Unassigned	
6	NUL - Unassigned	
7	NUL - Unassigned	
8	NUL - Unassigned	
9	NUL - Unassigned	

Figure 60: Key configuration for UEXT sets

Match DN
[Help](#)

Click on Assign to import the following features
 Flexible Call Forward No Answer(FDN)
 Hunt DN - All Calls or internal calls for CFTA(HUNT)
 Network Class Of Service(NCOS)
 Trunk Group Access Restriction(TGAR)
 Restrictive Class Of Service(CTD,UNR,TLD,SRE,FRE,FR1,FR2)

Search Results 1 Records

	Phone Type	Terminal Number	DN ^	FDN	HUNT	NCOS	TGAR	CLS
☐	1150	080 0 00 01	10			0	1	CTD

Figure 61: Copy Class of Service from desktop telephone

Configuring Features and Keys

1. Log on to UCM.
2. Click the Call Server element to launch Element Manager.

3. Click **Phones** from the navigation pane.

The Search for Phones page appears.

Note:

The telephone and call server database must be kept synchronized at all times. Customer information must be available for a telephone to be configured (for example, superloop, customer).

4. From the Search for Phones page, navigate to the **Phones** heading and click **Add**.

The New Phones page opens.

5. Select **UEXT-FMCL-Universal Extension FMCL** from the telephone **Type** list.
6. Complete 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 shows 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. Complete the required details for Key 0 and Key 1 and click **Validate** to perform validation of general properties, features and keys.
 9. Click **Finish**.

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.

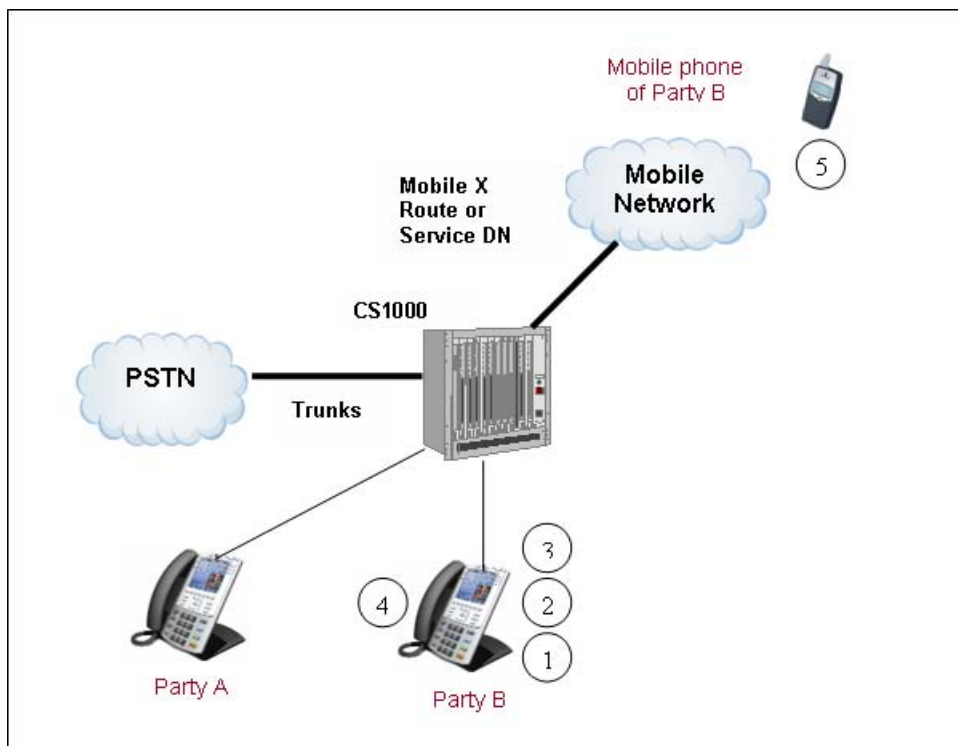


Figure 62: Handoff from Mobile telephone to Desktop telephone

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.

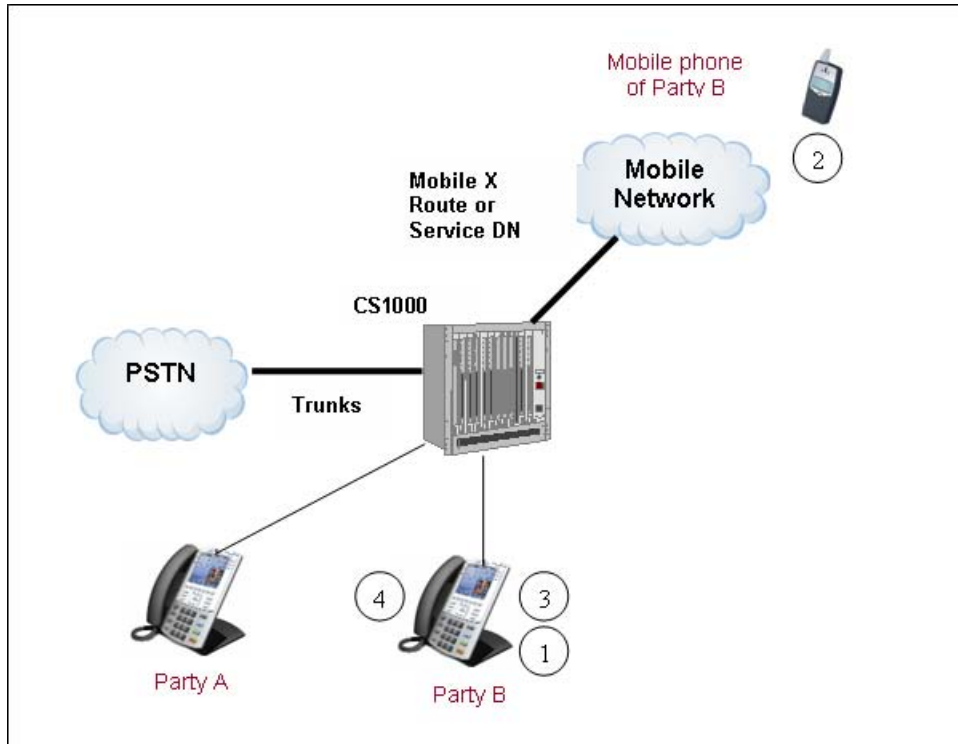


Figure 63: Handoff from Desktop telephone to Mobile telephone

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.

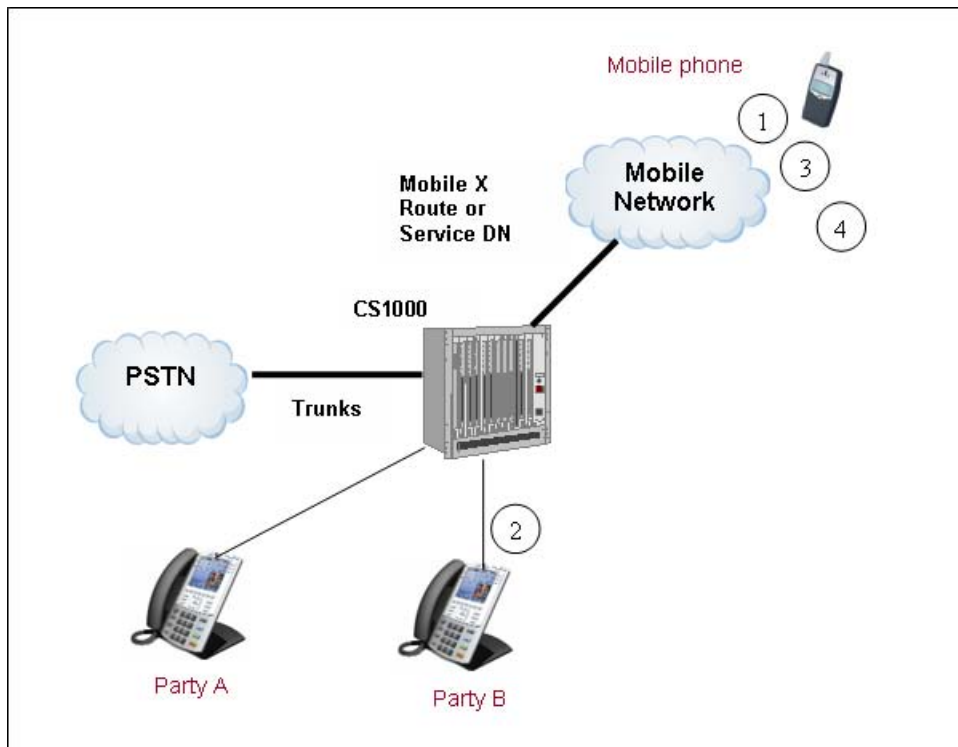


Figure 64: Call Transfer with consultation

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.

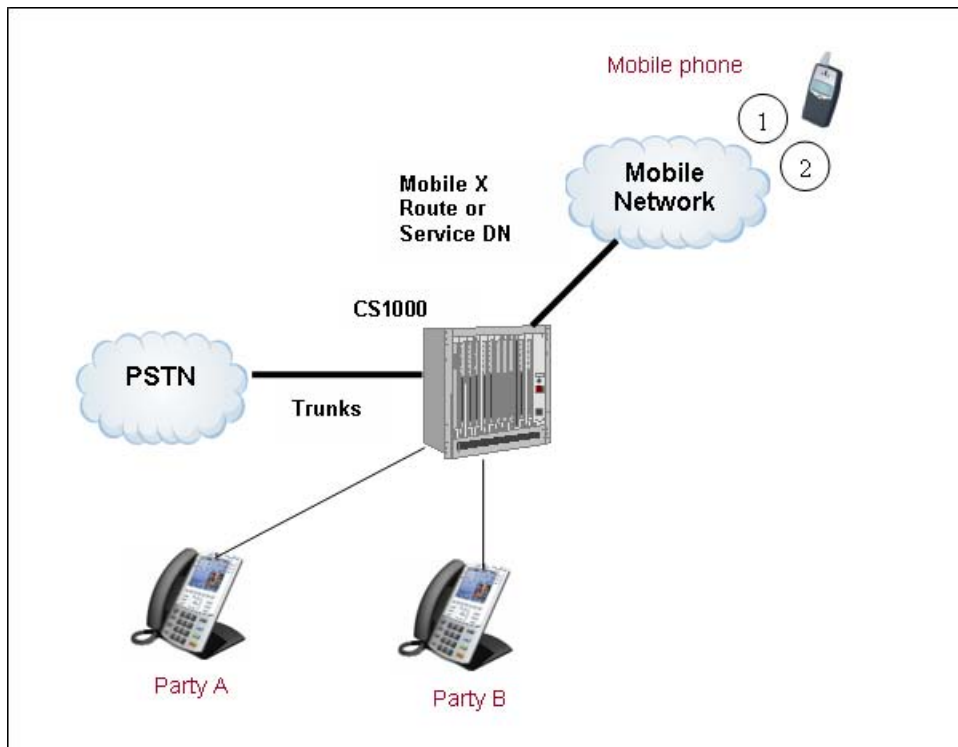


Figure 65: Blind Transfer

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.

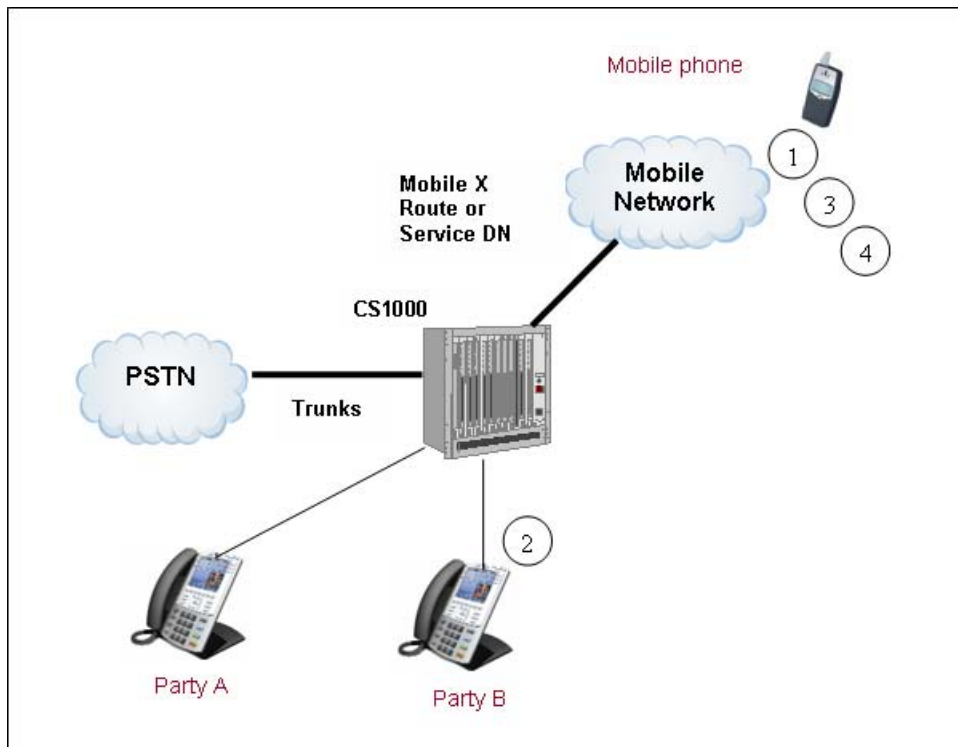


Figure 66: Conference with consultation

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.

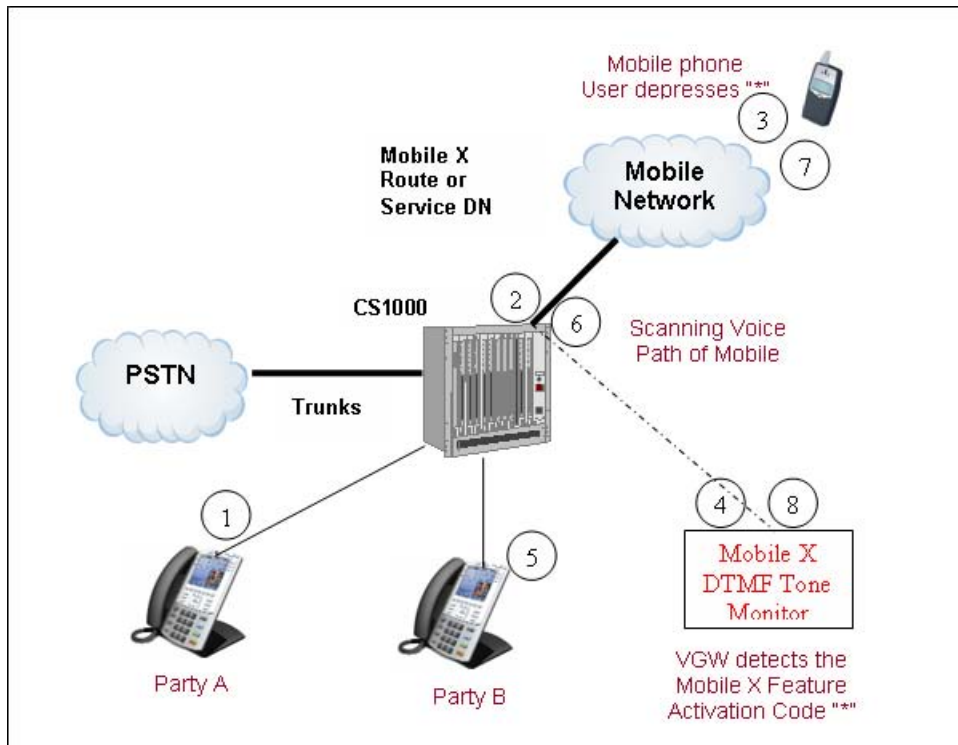


Figure 67: Feature activation example

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 68: Multi-language Messaging

Contents

This section contains information on the following topics:

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[Operating parameters](#) on page 556

[Feature interactions](#) on page 557

[Feature packaging](#) on page 557

[Feature implementation](#) on page 557

[Feature operation](#) on page 558

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 Avaya publications 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 System installation 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 190: 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 69: Multi-Party Operations

Contents

This section contains information on the following topics:

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[Feature interactions](#) on page 564

[Feature packaging](#) on page 574

[Feature implementation](#) on page 574

[Feature operation](#) on page 579

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.

Branch Office

In a Branch Office (BO) or Survivable Remote Gateway (SRG) environment, the RGNA prompt in the Customer Data Block is used when an external trunk is transferred on ringing and the called party, which is registered to the Main Office (MO), does not answer. Called parties that are not registered to the MO are routed according to local BO/SRG call modification configuration.

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 191: LD 15](#) on page 574
Configure the Multi-Party Operations parameters in the Customer data block.
2. [Table 192: LD 10](#) on page 577
Assign Three party Service (TSA) Class of Service to telephones.
3. [Table 193: LD 56](#) on page 577
Configure Control Dial Tone and recall Tones and Cadences for analog (500/2500-type) telephones and Meridian 1 proprietary telephones.

Table 191: 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
- - AOCS	(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.
	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.
- MHL D	(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
- AFNO - - ACNS	(NO), YES	YES – Alternative treatment is applied to Consultation calls where one of the parties disconnects. (Manual) Forced Camp-On Automatic. 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 192: 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 193: LD 56](#) on page 577to

Table 193: 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 *Avaya 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 telephones and digital telephones (M2000 series and Avaya 3900 Series Digital Deskphones).

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 194: 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 of [1](#) on page 583 (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 the 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 585. 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 70: Multi-Party Operations Enhancements

Contents

This section contains information on the following topics:

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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 195: 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 *Avaya 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 71: Multiple Appearance Directory Number Redirection Prime

Contents

This section contains information on the following topics:

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[Feature implementation](#) on page 605

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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 I s c u

- nnnn = the DN associated with the MARP TN
- I 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 (I 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 I s c u

- nnnn = the DN associated with the MARP TN
- I 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 (I s c u) for the telephone
- y = new TN (I s c u) for the telephone
- The following message appears again for each changed TN:

SCH5524 DN nnnn NEW MARP I s c u

- nnnn = the DN associated with the MARP TN
- I s c u = the new MARP TN assigned to DN nnnn

Automatic Call Distribution

Automatic Call Distribution (ACD) DN's are not assigned MARP TNs. Agent Individual DN's (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 68: CFW and MARP](#) on page 601.

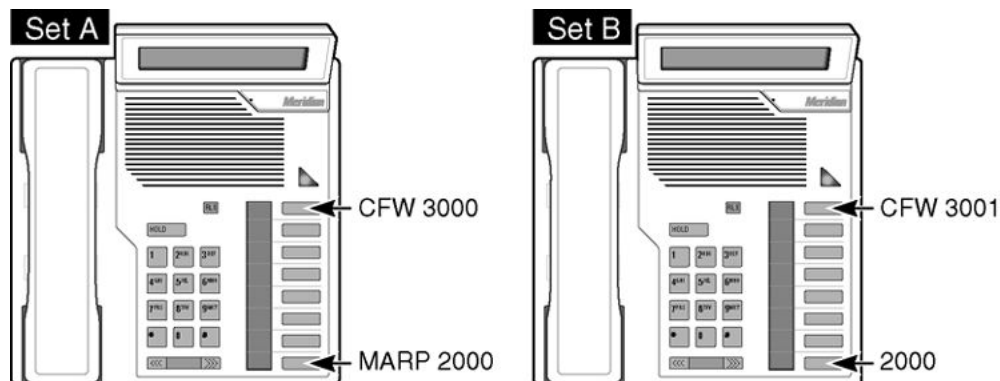


Figure 68: CFW and MARP

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 69: MARP control](#) on page 602.

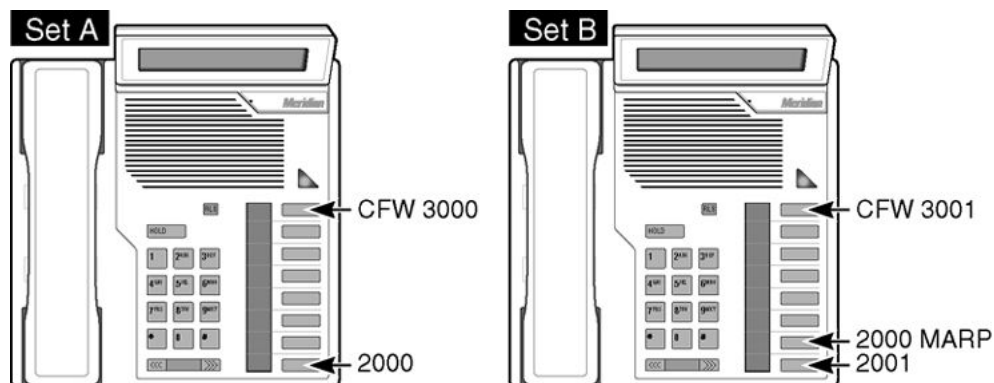


Figure 69: MARP control

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 DN's 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 196: LD 17](#) on page 606
Activate or deactivate MARP.
2. [Table 197: LD 10](#) on page 606
Add an analog (500/2500-type) telephone with a Single Appearance DN.
3. [Table 198: LD 10](#) on page 606
Add an analog (500/2500-type) telephone with a Multiple Appearance DN.
4. [Table 199: LD 10](#) on page 607
Change an analog (500/2500-type) telephone with a Multiple Appearance DN.
5. [Table 200: LD 11](#) on page 607
- Add a telephone with a Single Appearance DN.
6. [Table 201: LD 11](#) on page 607
Add a telephone with a Multiple Appearance DN.
7. [Table 202: LD 11](#) on page 608
Change a telephone with a Multiple Appearance DN.
8. [Table 203: LD 10](#) on page 608
Remove a MARP TN.
9. [Table 204: LD 11](#) on page 609
Remove a MARP TN.
10. [Table 205: LD 20 or LD 22](#) on page 609
Print MARP information.

If MARP is not activated, the overlays listed have this message printed: "MARF 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 196: 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 197: LD 10

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	500	500/2500 telephone.
TN		Terminal number
	I s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where I = loop, s = shelf, c = card, u = unit.
DN	xxx...x	Directory Number.
- MARP		MARP prints on the next line indicating this TN is the MARP for DN xxxx.

Table 198: LD 10

Prompt	Response	Description
REQ:	NEW	Add new data to the system.
TYPE:	500	500/2500 telephone.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
DN	xxx...x	Directory Number.

Prompt	Response	Description
- MARP ON TN	I s c u	MARP ON TN I s c u prints on the next line indicating TN I s c u is the current MARP.
- MARP	(NO) YES	(Do not) set the MARP to this new TN.

Table 199: LD 10

Prompt	Response	Description
REQ:	CHG	Modify existing data.
TYPE:	500	500/2500 telephone.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
DN	xxx...x	Directory Number.
- MARP ON TN	I s c u	This message indicates the current MARP is TN I s c u.
- MARP	(NO) YES	(Do not) set the MARP to this TN.

Table 200: 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
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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.
- MARP		MARP prints on the next line indicating this TN is the MARP for DN yyyy.
KEY		Reprompts until <CR> is entered.

Table 201: LD 11

Prompt	Response	Description
REQ:	NEW	Add new data to the system.

Prompt	Response	Description
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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	I s c u	MARP ON TN I s c u prints on the next line indicating TN I s c u is the current MARP.
- MARP KEY	(NO) YES	(Do not) set the MARP to this new TN. Reprompts until <CR> is entered.

Table 202: 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
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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.
- MARP ON TN	I s c u	MARP ON TN I s c u prints on the next line indicating TN I s c u is the current MARP.
- MARP KEY	(NO) YES	(Do not) set the MARP to the working TN. Reprompts until <CR> is entered.

Table 203: LD 10

Prompt	Response	Description
REQ:	OUT	Remove data from the system.
TYPE:	a...a	Telephone type.
TN		Terminal number

Prompt	Response	Description
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.

Table 204: 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
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.

Table 205: 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 72: Multiple Appearance Directory Number

Contents

This section contains information on the following topics:

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[Feature interactions](#) on page 613

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[Feature implementation](#) on page 621

[Feature operation](#) on page 621

Feature description

With the Multiple Appearance Directory Number (MADN) feature, a DN can be configured to appear on more than one telephone. You can have up to 30 appearances of the same DN. These appearances can be on single-line or multiline telephones.

The following multiple appearance options are provided for multiline telephones:

- Multiple Call Arrangement with Ringing (MCR)
- Multiple Call Arrangement without Ringing (MCN)
- Single Call Arrangement with Ringing (SCR)
- Single Call Arrangement without Ringing (SCN)

A DN can have a Multiple Call Arrangement (MCR or MCN) or a Single Call Arrangement (SCR or SCN). On multiline telephones, you can specify Multiple Call Ringing (MCR) or Multiple Call Non-ringing (MCN) options for each appearance of a DN. The MADN for MCR can be configured with the Multiple Key Ring All (MKRA) or Multiple Key Ring Lowest (MKRL) options. On single-line telephones, if a Multiple Call Arrangement DN is configured, the Multiple Call Ringing Allowed (MCRA) class of service must be assigned.

Telephones configured with MADN can be assigned to different loops if the Multiple Loop Directory Number (MLDN) enhancement is allowed in LD 17 using the prompt MLDN. However,

if an MADN telephone rings, the system reserves a connection for one appearance of the DN. If the MLDN prompt is configured to YES, other appearances can be blocked from being answered if the loop is busy at the time the telephone is answered.

A Multiple Appearance, Multiple Call Arrangement is available between multiline telephones, or with single-line telephones if they are configured as MCRA. You are allowed the same number of active calls as there are appearances of the DN. If the ring option is selected, the DN rings when an incoming call is directed to the idle DN. If the no ring option is selected, the DN appearance does not ring but rather the lamp associated with the DN flashes when an incoming call is directed to the DN.

In a Multiple Call Arrangement, the same DN can appear more than once on the same telephone. In a Multiple Call Arrangement, if MKRA is configured on the Customer Data Block, all idle DN keys on a MADN telephone are presented (or flash if configured as MCN) during an incoming call. If MKRL is configured, the lowest idle MCR/MCN DN key on each MADN telephone is placed in a ringing state during an incoming call. Other MCR/MCN keys on the same telephone remain idle and calls can be made or received regardless of the status of other MCR/MCN keys for the same TN. The same DN can be used for all the MCR/MCN keys which eliminates the necessity of maintaining numerous DN numbers for every secondary key. In situations where the system is upgraded from an earlier release, the default value is set to MKRA. If a new default Customer (CUST) block is created, the default value is MKRL.

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.

If the ring option is selected, a ringing tone accompanies lamp flashing during an incoming call to a DN. Privacy is inherent in active calls, except in a mixed arrangement, for example, analog (500/2500-type) telephones and multiline telephones with an appearance of the same DN. For more information, see [Privacy](#) on page 619, [Privacy Override](#) on page 619, and [Privacy Release](#) on page 620.

If the Multiple Appearance Redirection Prime (MARP) feature is enabled in the Configuration Record, call redirection parameters such as Hunt and Call Forward No Answer (CFNA) are taken from the TN data block of the appearance of the MADN that is designated as the MARP. If the MARP appearance for each DN was designated prior to enabling the feature in the Configuration Record, it has no effect. If the MARP feature is disabled, call redirection parameters such as Hunt and CFNA are derived from the TN data block (LD 20 TNB) of the primary appearance of the called DN. If there is more than one primary appearance, the parameters are selected from the last TN in the DN block for the DN (LD 22 DNB). For more information, see [Call Forward by Call Type, Call Forward No Answer, Second Level](#) on page 614.

Feature interactions

Automatic Redial

An Automatic Redial (ARDL) call from a Single Call Ringing (SCR) or Single Call Non-Ringing (SCN) configuration is redialed only when all the telephones that have the same DN are free.

An ARDL call from a Multiple Call Ringing (MCR) or Multiple Call Non-Ringing (MCN) configuration is redialed when the originating key becomes free.

Automatic Wake Up

With the Automatic Wake Up (AWU) feature configured, all Multiple Appearance DNs (MADN) ring, including both primary and secondary DNs. The wake up request is programmed using the Wake Up key and applies to telephones that have the primary DN on key 0. The Wake Up indicator operates as described on telephones where the wake up request is programmed.

After the wake up programming sequence is finished, all Wake Up indicators on MADNs are updated. If two or more MADN telephones are programmed for a wake up request at the same time, the telephone that is last saved overrides the previous request. Telephones with the same primary DN get the request time of the telephone that last programs a request. If the last telephone cancels the request, all requests are canceled.

If the AWU Recall option is chosen, the recall is presented to any idle attendant console in the same Console Presentation Group (CPG) that is equipped with the AWU key.

Automatic Wake FFC Delimiter

For Multiple Appearance Directory Number (MADN), wake up information is stored, deleted, and queried from a DN first primary appearance Terminal Number (TN).

Call Detail Recording on Redirected Incoming Calls

If a call is forwarded and the Directory Number (DN) of the telephone is a Multiple Appearance Directory Number (MADN), the Terminal Number (TN) of the telephone prints out in the AUX ID field, for example, line two of the Call Detail Recording (CDR) record.

Call Forward by Call Type, Call Forward No Answer, Second Level

If the Multiple Appearance Redirection Prime (MARF) feature is enabled in the Configuration Record, call redirection parameters such as Call Forward No Answer (CFNA) are derived from the TN data block of the appearance of the Multiple Appearance Directory Number (MADN) that is designated as the MARF. If there is more than one primary appearance, the parameters are selected from the last TN of the DN block.

Prior to configuring Call Redirection parameters for MADN telephones where more than one primary appearance of a MADN exists, consider the following.

- 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 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 list.
 - If a telephone undergoes a Service Change, the TN moves to the beginning of the DN list, regardless of the numerical value of the TN. The TN remains at the beginning of the list until another service change or a SYSLOAD.
 - If a DN is assigned as the primary DN on one telephone, and as a secondary DN on one or more telephones, the DN list is organized as described in the preceding bullet. However, if only one primary appearance of a DN exists, call redirection parameters are derived from the TN of the primary appearance telephone, even if the end of the list. A primary 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 multiline telephones, the analog (500/2500-type) telephones are listed in numerical TN order at the top of the list. Multiline 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 to the beginning of the list. A Service Change to a multiline telephone moves the TN to the end of the list.
 - A SYSLOAD restores the list back to numerical TN order, with analog (500/2500-type) telephones at the top and multiline telephones at the bottom. Call Redirection parameters continue to be derived, as described in the preceding paragraphs.

Note:

It is not necessary to change any data to register a service change activity. To move a telephone to the end of the list, open the service change data, and accept the default data settings.

Call Forward, Remote (Attendant and Network Wide)

The Call Forward, Remote (RCFW) feature applies to the primary appearances of Multiple Appearance DNs (MADN). Avaya recommends that you configure only one appearance of a Multiple Appearance DN as the primary DN.

For the case of multiple stations with the same primary DN and Station Control Password (SCPW), the RCFW operation applies to the station that has the Multiple Appearance Redirection Prime (MARF) assigned to it. If the stations with the assigned DNs and SCPW are not configured as the MARF TN for that DN, the RCFA and RCFD applies to all stations that match the DN and SCPW.

The attendant-based RCFW feature applies the RCFW operation to the Primary DN with MARF status. If the DN is not the Primary DN or does not have MARF status, the user receives an overflow tone.

Calling Party Name Display Denied

For a call placed to a Multiple Appearance Directory Number (MADN) telephone, you can suppress the name on the display for the called telephone by configuring any of the Terminal Numbers (TN) with NAMF Class of Service. The digit display on the called telephone cannot be suppressed; the digits display even though the Class of Service on any of the Terminal Numbers is DIGF. 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 called telephone display is subject to the Class of Service configured for the answering telephone. The answering telephone display 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 that apply to single and primary appearance DNs of MADNs.

China - Attendant Monitor

If the Attendant Monitor monitors a Multiple Appearance DN, the Multiple Appearance Redirection Prime (MARF) TN is the monitored TN.

Controlled Class of Service

Controlled Class of Service (CCOS) restriction levels are activated or canceled on controlled telephones through the Primary Directory Number (PDN). If the PDN of a multiline telephone activates CCOS, all DNs on that telephone are restricted. If the DN is a PDN on another telephone with CCSA Class of Service, the telephone is restricted.

Controlled Class of Service, Enhanced

All Controlled Class of Service (CCOS) restriction levels are activated and canceled from the Primary Directory Number (PDN) for CCOS controlling telephones. The PDN for an SL-1 telephone becomes CCOS active, and all DNs are restricted for that telephone and other telephones where the DN is a PDN (with CCSA Class of Service).

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

If the attendant attempts to extend a call to a DN that appears on more than one telephone, the DN can be one of the following:

- Multiple Call Arrangement with Ringing (MCR): The idle stations ring for calls that terminate on an MCR DN. The call is established to the station that answers first. All other stations remain idle.
- Multiple Call Arrangement with No Ringing (MCN): The idle stations DN key flash for calls that terminate on an MCN DN. The call is established to the station that answers first. All other stations remain idle.
- Single Call Arrangement with Ringing (SCR): The idle stations ring for calls that terminate on an SCR DN. The call is established to the station that answers first. All other stations are busy.
- Single Call Arrangement with No Ringing (SCN): The idle stations DN key flash for calls that terminate on an SCN DN. The call is established to the station that answers first. All other stations remain idle.

Digital Trunk Interface (DTI) - Commonwealth of Independent States (CIS)

Two stations with the same Multiple Appearance Directory Number (MADN) can be assigned different ANI categories because the ANI category is defined on a per telephone basis

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 multiline telephone data block (LD 11), the analog (500/2500-type) telephone data block (LD 10), or Attendant Administration.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The locked or unlocked state applies to all Terminal Numbers (TN) with the same primary DN and Station Control Password (SCPW). TNs 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 telephones 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 the number of DN appearances.

Group hunt

Avoid Multiple Appearance DN (MADN) multiple call arrangements in a group hunt list when MADN Single Call Arrangements are treated the same as Single Appearance DNs.

With MADN Multiple Call Arrangement, the idle or busy status of the MADN is determined by the terminal number (TN) data block of the primary appearance of the called DN. If there is more than one primary 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). The hunt cycle regards the idle appearances of the MADN as busy and attempts to terminate on the next idle member of the group hunt list. If a MADN Multiple Call Arrangement is used, a supervisor telephone must be assigned to the hunt group. The supervisor telephone is given the primary appearance of the MADN. Other appearances are MADN programmed as a secondary DN (any DN key other than 0). The supervisor telephone controls the status of the MADN and 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 Multiple Appearance Directory Number (MADN) Redirection Prime (MARF) Terminal Number (TN). If the MARF system option is disabled, Hunting proceeds as if MARF did not exist.

If all the telephones in the MADN group are multiline telephones, ringing telephones are placed at the top of the DN list, and non-ringing telephones are placed at the bottom.

If a MADN 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 the multiline 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 multiline telephone moves it to the bottom of the list. Call redirection follows the TN order from top to bottom.

The MARF TN is always checked to determine if and how the call is to be redirected by Hunting, regardless of where the MARF TN appears in the TN list of the DN block. You do not need to search the TN list of the DN block. Hunting follows the hunt chain based on the originally dialed DN. The function and requirements for Hunting are not changed by the MARF feature. The basic change introduced by the MARF feature is to have a designated TN, the MARF TN, as the TN supplying the call redirection parameters.

If the MARF TN does not have Hunting control enabled, no Hunting is attempted. Other features for redirecting calls to busy DNs can be attempted based on the MARF TN.

A Short Hunting sequence begins when the MARF TN of a busy DN performs Short Hunting. When a Short Hunt begins, it completes on the existing telephone before going to the Hunt DN. The precedence of Short Hunting over normal Hunting is maintained. After 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. A Hunt Chain connects Short Hunting sequences only through Hunt DNs.

Last Number Redial

A last number dialed on a Directory Number (DN) with multiple appearances is stored 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 are defined as Single Call Ringing (SCR) keys or as Multiple Call Ringing (MCR) keys. For SCR DN keys on MADN telephones, only one call can be answered at a time. For example, after a call is answered, future calls to that DN receive a busy tone until the DN is disconnected.

For MCR DN keys, a caller only receives a busy tone after every DN is busy. If one call taker is answering a call and there are other call takers available, the call is directed to the telephones of the available call takers. Any available call taker can answer the new call.

Message Registration

For Multiple Appearance Directory Number (MADN), the system selects the appropriate meter for the DN by assessing the meter of the most recently configured telephone with a Primary DN 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 multiline telephones, the Privacy feature is in place. A call must be placed on Hold or Privacy Release is activated for another appearance to enter the call. Privacy does not apply for any appearance of the DN if there is a shared configuration between multiline and single-line telephones. If a DN is shared, a call can be entered 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 the following calls:

- Multiple Appearance
 - Multiple Call Arrangement with Ringing (MCR)
 - Multiple Call Arrangement without Ringing (MCN)

Remote Call Forward

With a Multiple Appearance Directory Number (MADN), if both telephones have a Station Control Password (SCPW), Remote Call Forward does not always operate as intended. For example, if Call Forward has been activated using the Remote Call Forward feature, you cannot deactivate Call Forward from the same telephone on which it is active.

Three Wire Analog Trunk - Commonwealth of Independent States (CIS)

Two stations with the same Multiple Appearance Directory Number (MADN) can be assigned different ANI categories as the ANI category is defined on a per telephone basis for Three Wire Analog Trunks.

Voice Call

If a Voice Call DN is added to a second telephone, the DN becomes a Multiple Appearance Directory Number (MADN). Voice Call does not support MADN.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 206: 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	Terminal Number for Large Systems.
KEY	xx MCN yyy...y	Add a multiple-call non-ringing DN key, where: xx = key number, and yyy...y = DN.
	xx MCR yyy...y	Add a multiple-call ringing DN key, where: xx = key number, and yyy...y = DN.
	xx SCN yyy...y	Add a single call non-ringing DN key, where: xx = key number, and yyy...y = DN.
	xx SCR yyy...y	Add a single call ringing DN key, where: xx = key number, and yyy...y = DN.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 73: Multiple Console Operation

Contents

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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 207: LD 12](#) on page 626
Select attendant console number.
2. [Table 208: LD 15](#) on page 626
Select Supervisory Console.
3. [Table 209: LD 93](#) on page 626
Configure Multi-Tenant Service.

Table 207: LD 12

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
...		
ANUM	1-63	Attendant Number.

Table 208: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
...		
SPVC	(0)-63	Supervisory Console.

Table 209: 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 74: Multiple Customer Operation

Contents

This section contains information on the following topics:

[Feature description](#) on page 629

[Operating parameters](#) on page 629

[Feature interactions](#) on page 629

[Feature packaging](#) on page 630

[Feature implementation](#) on page 630

[Feature operation](#) on page 630

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 75: Multi-Site Mobility Networking

Contents

This section contains information on the following topics:

[Feature description](#) on page 631

[Operating parameters](#) on page 632

[Feature interactions](#) on page 632

[Feature packaging](#) on page 633

[Feature implementation](#) on page 633

[Feature operation](#) on page 637

Feature description

Multi-Site Mobility Networking (MSMN) allows a 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 Avaya 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 Avaya 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 210: LD 10](#) on page 634
Add a new DCS handset.
2. [Table 211: LD 10](#) on page 635
Copy DCS handsets on DMC.
3. [Table 212: LD 10](#) on page 635
Remove DCS handsets.
4. [Table 213: LD 10](#) on page 636
Convert handset type 500 to DCS.
5. [Table 214: LD 20](#) on page 636
Print actual DMC TN and Phantom DMC TN list.
6. [Table 215: LD 81](#) on page 636
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 *Avaya DECT Fundamentals*, NN43120-114.

Table 210: 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.
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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	I s c	DECT Mobility Controller Location. Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where I = loop, s = shelf, 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 211: 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	I s c	DECT Mobility Controller Location. Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card.

Table 212: 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	I s c	DECT Mobility Controller Location. Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card.

Table 213: 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 I 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 214: LD 20

Prompt	Response	Description
REQ	PRT	Request.
TYPE	DCS	Digital Cordless Set.
TN		Terminal Number for actual DMC.
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
		Phantom Terminal Number.
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, 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 215: LD 81

Prompt	Response	Description
REQ	LST	Request.
FEAT	VSIT	Feature Request - DECT visitors.

Prompt	Response	Description
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 76: Multi-Tenant Service

Contents

This section contains information on the following topics:

[Feature description](#) on page 639

[Operating parameters](#) on page 645

[Feature interactions](#) on page 646

[Feature packaging](#) on page 657

[Feature implementation](#) on page 657

[Feature operation](#) on page 661

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 70: Tenant-to-Tenant access](#) on page 641, 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.

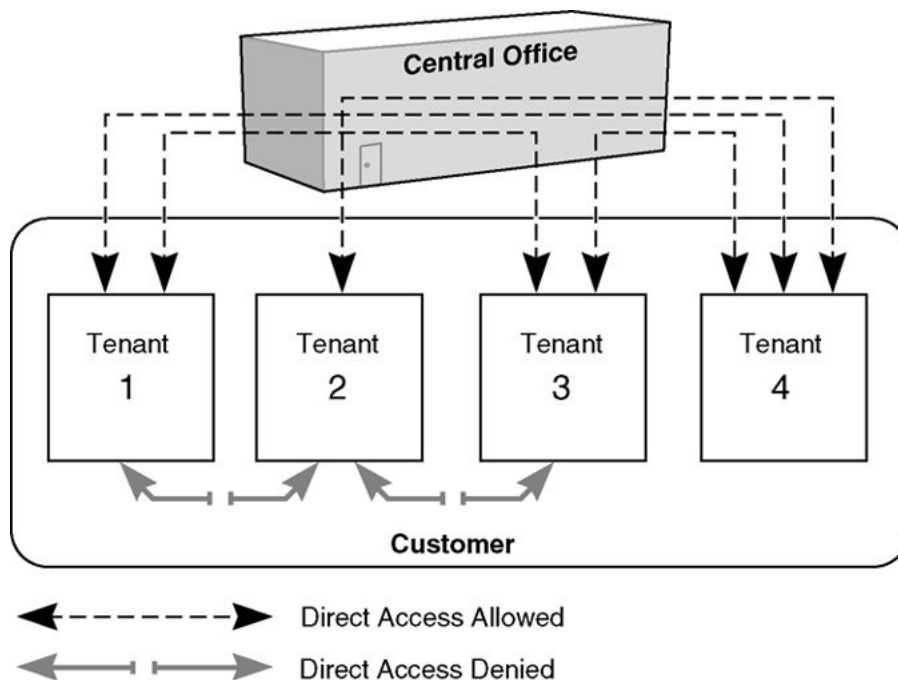


Figure 70: Tenant-to-Tenant access

As shown in [Table 216: Tenant-to -Tenant Access allowed or denied](#) on page 641, Tenant-to-Tenant Access allows or denies tenants of the customer:

Table 216: 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 71: Tenant-to-Trunk Route Access](#) on page 642 shows a diagram of the following tables.

Table 217: 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 71: Tenant-to-Trunk Route Access](#) on page 642, Tenants 2 and 3 share access to route 2.

Table 218: 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

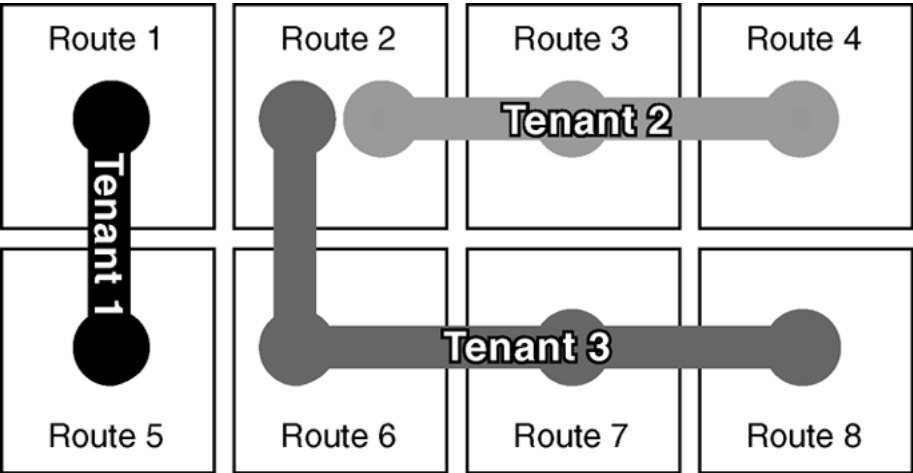


Figure 71: Tenant-to-Trunk Route Access

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 643.

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 219: Typical attendant group arrangement](#) on page 644, 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 219: Typical attendant group arrangement](#) on page 644, incoming calls on route 2 are automatically presented to Attendant Console Group 5 (console 7 only).

Table 219: 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

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 220: CDR record types and descriptions](#) on page 647.

Table 220: 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 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.

Ring Number Pickup

Ring 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 221: LD 93](#) on page 658
Enable, disable, or print Multi-Tenant Service for a specified customer.
2. [Table 222: LD 93](#) on page 658
Allow, deny, or print Tenant-to-Tenant Access for a specified tenant.
3. [Table 223: LD 93](#) on page 658
Allow, deny, or print Tenant-to-Route Access for a specified trunk route.
4. [Table 224: LD 93](#) on page 659
Add Console Presentation Group.
5. [Table 225: LD 93](#) on page 659

Assign Tenant-to-Attendant Console access.

6. [Table 226: LD 93](#) on page 659

Assign Attendant Console group number.

7. [Table 227: LD 10](#) on page 660

Add Multi-Tenant Service assignments on analog (500/2500) telephones.

8. [Table 228: LD 11](#) on page 660

Add Multi-Tenant Service assignments on proprietary telephones.

Table 221: 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 222: 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 ALLOW	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).
ALLOW	1-511 1-511 ALL	Tenant numbers allowed access to and from this tenant (prompted if ACC = ALLOW). All tenant numbers allowed access to and from this tenant.

Table 223: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
ACC	DENY ALLOW	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.
ALLOW	1-511 1-511 ALL	Tenant numbers allowed access to this route (prompted if ACC = ALLOW). All tenant numbers allowed access to this route

Table 224: 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 225: 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 226: LD 93

Prompt	Response	Description
REQ	CHG, PRT	Change, or print.

Prompt	Response	Description
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.
AGNO	0-63	Attendant Console group number.

Table 227: 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.
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 228: 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.
CLS	(TEND) TENA	Tenant service (denied) (station shares customer resources and is a non-tenant). Tenant service allowed.
TEN	1–511	Tenant number. Tenant 0 is reserved for telephones with a TEND Class of Service. Prompted if CLS = TENA.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 77: Music

Contents

This section contains information on the following topics:

[Feature description](#) on page 663

[Operating parameters](#) on page 664

[Feature interactions](#) on page 665

[Feature packaging](#) on page 666

[Feature implementation](#) on page 667

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 *Avaya 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 671).

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 229: LD 17](#) on page 667
Enable conference loops for Music on Hold.
2. [Table 230: LD 16](#) on page 667
Enable a music route.
3. [Table 231: LD 14](#) on page 668
Enable a music trunk.
4. [Table 232: LD 16](#) on page 668
Enable Music on Hold for trunk routes.
5. [Table 233: LD 23](#) on page 669
Enable Music for an Automatic Call Distribution Directory Number.

Table 229: 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 230: 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 231: LD 14

Prompt	Response	Description
REQ	NEW	New.
TYPE	MUS	Music trunk.
TN		Terminal number
	l s c u	Format for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, 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.
CFLP	0-158	Conference loop assigned to music in LD 17.

Table 232: 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.
TKTP	COT DID FEX TIE WAT	Route type.
MUS	(NO) YES	Music on Hold (is not) or is to be provided for this trunk route.
MRT	xxx	Music route number.

Table 233: 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 78: Music, Enhanced

Contents

This section contains information on the following topics:

[Feature description](#) on page 671

[Operating parameters](#) on page 672

[Feature interactions](#) on page 672

[Feature packaging](#) on page 673

[Feature implementation](#) on page 673

[Feature operation](#) on page 676

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 234: Features vs. no Music, Music, and Enhanced Music](#) on page 671.

Table 234: 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 663.

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 235: LD 17](#) on page 674
Add or change Conference loops for Music on Hold.
2. [Table 236: LD 15](#) on page 674

Enable Music Customer Data Block.

3. [Table 237: LD 16](#) on page 674

Enable a Music route.

4. [Table 238: LD 14](#) on page 675

Enable a Music trunk. At least one Music trunk per network group is required for each customer requiring Music.

5. [Table 239: LD 16](#) on page 675

Enable Music on Hold for trunk routes.

Table 235: 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 236: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CDB FTR	Customer Data Block Features and options
CUST	0-99	Customer number Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system.
- MUS	(NO) YES	Enhanced music for telephones.
- MUSR	0-511	Music route for telephones.

Table 237: 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 238: LD 14

Prompt	Response	Description
REQ	NEW CHG	New Change
TYPE	MUS	Music trunk.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
CFLP	0-158	Conference loop assigned to music in LD 17.

Table 239: 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 79: Music Broadcast

Contents

This section contains information on the following topics:

[Feature description](#) on page 677

[Operating parameters](#) on page 680

[Feature interactions](#) on page 681

[Feature packaging](#) on page 682

[Feature implementation](#) on page 683

[Feature operation](#) on page 684

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 72: License header in LD 14](#) on page 679 is an example of the updated header:

TNS	AVAIL: xxxxx	USED: xxxxx	TOT: xxxxx
MUS CON	AVAIL: xxxxx	USED: xxxxx	TOT: xxxxx

Figure 72: 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 *Avaya 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 73: TFC111 Report for a broadcasting music route](#) on page 680 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 73: TFC111 Report for a broadcasting music route

Operating parameters

Music Broadcast requires any Music trunk and an external music source or an Avaya 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 240: LD 16](#) on page 683
Change an existing non-broadcasting Music route to a broadcasting Music route.
2. [Table 241: LD 16](#) on page 684
Enable Conference-based Music route.
3. [Table 242: LD 14](#) on page 684
Configure Conference-based Music trunks.

Music Broadcast

Table 240: 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 Avaya Communication Server 1000E (Avaya CS 1000E) system.
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 241: LD 16

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
ROUT	0-511	Route number for Large 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 242: LD 14

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	MUS	Music trunk.
TN		Terminal number
	I s c u	Format for Large System and CS 1000E system, where I = loop, s = shelf, c = card, u = unit.
...		
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
...		
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 80: Music on Hold, Set-based

Contents

This section contains information on the following topics:

[Feature description](#) on page 685

[Operating parameters](#) on page 686

[Feature interactions](#) on page 686

[Feature packaging](#) on page 686

[Feature implementation](#) on page 686

[Feature implementation](#) on page 686

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 243: 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 244: 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.

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