Features and Services Fundamentals — Book 6 of 6 (S to Z) Avaya Communication Server 1000

7.5
NN43001-106, Standard 05.06
August 2012

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Contents

Chapter 1: New in this release

The following sections detail what is new in this document for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.5.

Features

There are no updates to the feature descriptions in this document.

Other changes

See the following section for information about changes that are not feature-related.

Revision History

August 2012	Standard 05.06. This document is up-issued to reflect changes in technical content for the “Trunk to Trunk Connections” feature interaction with Conference. Information on Variable Flash Timing and Ground Button removed as hardware required is no longer available.
December 2011	Standard 05.05. This document is up-issued to support the removal of End of Life (EoL) and Manufactured Discontinued (MD) hardware content and associated diagrams.
September 2011	Standard 05.04. This document is up-issued to reflect changes in technical content for the feature Virtual Office-only IP Phones.
March 2011	Standard 05.03. This document is up-issued to support Avaya Communication Server 1000 Release 7.5.
February 2011	Standard 05.02. This document is up-issued to remove legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.
November 2010	Standard 05.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.5.
June 2010	Standard 04.02. Up-issued to reflect changes in technical content.

June 2010	Standard 04.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.0.
June 2009	Standard 03.03. This document is up-issued to support Communication Server 1000 Release 6.0.
September 2008	Standard 02.06. This document is up-issued to add reference for information on M3900 for CS Release 5.5.
May 2008	Standard 02.05. This document is up-issued to support Communication Server Release 5.5.
December 2007	Standard 02.04. This document is up-issued to support Communication Server Release 5.5.
July 2007	Standard 01.04. This document is up-issued to revise the 500 Telephone Features and Bandwidth Management Support for Network Wide Virtual Office chapters in Book 1 and Conference Warning Tone Enhancement chapter in Book 2).
June 2007	Standard 01.03. This document is up-issued to revise the Software Licenses chapter in Book 6.
June 2007	Standard 01.02. This document is up-issued to revise 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) This document also includes the following updates: • Corrections to Trunk Route Optimization - Before Answer on page 534 (Book 5) and to Trunk Route Optimization - Before Answer on page 540 (Book 5). • Updated the description of EXTT prompt in LD 15 on page 338 (Book 6).
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). • Addition of M3900 Set-to-Set Messaging feature on pages 801 to 806 (Book 2). • Addition of M3900 series digital telephone feature reference on pages 341, 342 of the Personal Directory chapter (Book 3).

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). • Addition of IPMG on CS1000E to the following: operating parameters on page 364 (Book 3); and LD 97 on page 379 (Book 3). • 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). • Addition of Flexible Feature Codes to list on pages 371 to 376 of Flexible Feature Codes chapter (Book 2). • Correction to Message Intercept for Set Status Lockout on pages 982-983 (Book 2). • Correction to SECA001 alarm message on page 402 (Book 1).
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). • Addition of IP Phones to supported sets referenced in Selectable Conferee Display and Disconnect on pages 667 to 700 (Book 3).
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 3 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 documents. 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 documents have been retired to reflect this change: 553-2611-200, 553-2751-104, 553-2831-100, 553-2721-111 and 553-2941-100.
December 1995	Issue 5.00. This is the Release 21.1x standard version of this document.
July 1995	Issue 4.00. This is the Release 21 standard version of this document.
October 1994	Issue 2.0. This is the Release 20.1x soak version of the document.
July 1994	Issue 1.0. This is the Release 20.0x standard version of this document.

Chapter 2: Customer service

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Chapter 3: Features and Software options

Package Name	Number	Mnemonic	Release
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• 16-Button Digitone/Multifrequency Operation			
2 Mbit Digital Trunk Interface	129	DTI2	10
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• 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			

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
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• 2500 Set Features			
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• Alternative Conference PAD Levels			
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• Alternative Loss Plan for China			
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Aries Digital Sets	170	ARIE	14

Package Name	Number	Mnemonic	Release
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• Meridian Modular Telephones			
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• Attendant Administration			
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Attendant Announcement	384	AANN	25.40
• Attendant Announcement			
Attendant Break-In/Trunk Offer	127	BKI	1
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• 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)			
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• Attendant Busy Verify			
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• Attendant Calls Waiting Indication			
• Attendant Consoles			
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Attendant Forward No Answer	134	AFNA	14
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• Attendant Interpositional Transfer			
• Attendant Lockout			
Attendant Overflow Position	56	AOP	1

Package Name	Number	Mnemonic	Release
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• 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			
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• 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)			
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Automatic Call Distribution Enhanced Overflow	178	EOVF	15
• Automatic Call Distribution Enhanced Overflow			
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• Automatic Call Distribution Load Management Reports			
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• Call Processor Input/Output)			
Automatic Call Distribution Package C	42	ACDC	1
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Package Name	Number	Mnemonic	Release
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Automatic Call Distribution Package D, Auxiliary Security	114	AUXS	12
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Automatic Call Distribution Package D	50	ACDD	2
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• Automatic Guard Detection			
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• Automatic Installation			
Automatic Line Selection	72	LSEL	4
• Automatic Line Selection			

Package Name	Number	Mnemonic	Release
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• Automatic Redial			
• Automatic Timed Reminders			
Automatic Wake-Up	102	AWU	10
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• Auxiliary Signaling			
• B34 Dynamic Loss Switching (see also packages 164 and 203)			
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• Background Terminal Facility			
Basic Alternate Route Selection	57	BARS	1
• Network Alternate Route Selection/Basic Alternate Route Selection Enhancement – Local Termination (see also package 58)			
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• Automatic Call Distribution Calls on Hold (see also package 131)			
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Package Name	Number	Mnemonic	Release
• Automatic Call Distribution in Night Service			
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Basic Call Processing	0	BASIC	1
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• Call Center Transfer Connect			

Package Name	Number	Mnemonic	Release
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• Call Detail Recording with Optional Digit Suppression			
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• 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			
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• Call Forward, Internal Calls			
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• Call Processor Input/Output)			
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• Call Waiting Notification (Meridian 911)			
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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Package Name	Number	Mnemonic	Release
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• Dial Pulse/Dual-tone Multifrequency Conversion			
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• Attendant Timed Reminder Recall and Attendant Third Party Service			
• Call Back when Free and Next Used			
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• 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	288	DPNSS_ES	21
• DPNSS1 Executive Intrusion			
Enhanced DPNSS1 Gateway	284	DPNSS189I	20
• Enhanced DPNSS1 Gateway			
Enhanced Hot Line	70	HOT	4/10
• Hot Line			
• Network Intercom			
• Enhanced input/output buffering			
• Enhanced Maintenance (Patching)			
Enhanced Music	119	EMUS	12
• Music, Enhanced			
Enhanced Night Service	133	ENS	20
• Enhanced Night Service			
• Enhanced package printout			
• Equal Access Compliance			
Euro ISDN Trunk - Network Side	309	MASTER	22
• EuroISDN Trunk - Network Side			
Euro ISDN	261	EURO	20
• ISDN – Advice of Charge for EuroISDN			
• ISDN BRI and PRI Trunk Access for Europe (EuroISDN)			
• EURO ISDN Continuation			
Euro Supplementary Service	323	ETSI_SS	22
• EuroISDN Call Completion Supplementary Service			
Executive Distinctive Ringing	185	EDRG	16
• Executive Distinctive Ringing			
FCC Compliance for DID Answer Supervision	223	FCC68	17
• Federal Communications Commission Compliance for DID Answer Supervision			

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	FNP	14
• Alternative Routing for DID/DOD			
• Flexible Numbering Plan			
• Special Dial Tones after Dialed Numbers			
• Flexible Numbering Plan Enhancement			
• Flexible Orbiting Prevention Timer			
Flexible Tones and Cadences	125	FTC	16
• Flexible Tone and Digit Switch Control			
• Reverse Dial on Routes and Telephones			
• Tones and Cadences			

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_VTRK	3.0
• IP Peer Networking Phase 2			
• Branch Office			
HiMail Fax Server	195	FAXS	18
History File	55	HIST	1
• History File			
Hold in Queue for IVR	218	IVR	18

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	140	DCON	15
• Digital Attendant Console			
M2317 Digital Sets	91	DLT2	9
• M2317 Digital Sets			
M3000 Digital Sets	89	TSET	7
• M3000 Telephones			
M3900 Full Icon Support	397	ICON_ PACKAGE	3.0
• M3900 Full Icon Support			
M3900 Phase III Virtual Office Enhancement	387	VIR_OFF_ ENH	25.40
• Virtual Office Enhancement			
M3900 Ring Again	396	M3900_RG A_PROG	3.0
M911 Enhancement Display	249	M911 ENH	25
• 10/20 Digit ANI on 911 Calls			
Maid Identification	210	MAID	17
• Maid Identification			
• Make Set Busy and Voice Call Override			
Make Set Busy	17	MSB	1
• Make Set Busy			
• Make Set Busy Improvement			
• Malicious Call Trace on Direct Inward Dialing			
Malicious Call Trace	107	MCT	10
• Enhanced Malicious Call Trace			
• Malicious Call Trace			
• Malicious Call Trace DN/TN Print			
• Malicious Call Trace Idle			
• Manual Line Service			
• Manual Service Recall to Attendant			
• Manual Signaling (Buzz)			
• Manual Trunk Service			

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

Package Name	Number	Mnemonic	Release
Meridian Link Modular Server	209	MLM	16
• Meridian Link Enhancements			
Meridian SL-1 ST Package	96	SLST	9
• Meridian SL-1 ST Package			
Message Intercept	163	MINT	15
• Message Intercept			
Message Waiting Center	46	MWC	1
• Message Waiting Lamp Maintenance			
• Message Waiting Unconditional			
Message Waiting Indication Interworking with DMS	219	MWI	19
• Message Waiting Indication (MWI) Interworking			
Mobile Extensions	412	MOBX	5.50
Modular Telephone Relocation			
Multifrequency Compelled Signaling	128	MFC	9
• China Number 1 Signaling Trunk Enhancements (see also packages 49, 113, and 131)			
• China Number 1 Signaling – Active Feature Dial Tone (see also package 126)			
• China Number 1 Signaling – Audible Alarm (see also package 126)			
• 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			
Multifrequency Signaling for Socotel	135	MFE	10
• Multifrequency Signaling for Socotel			

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_USER	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) • 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)	126	OPCB	14
Optional Features • 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)	1	OPTF	1
Optional Outpulsing Delay • Optional Outpulsing Delay	79	OOD	5

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_IPE	
• Remote Intelligent Peripheral Equipment			
Remote Virtual Queuing	192	RVQ	18
• Network Drop Back Busy and Off-hook Queuing (see also package 62)			
• Remote Virtual Queuing			
Resident Debug	82	RSDB	9
• Restricted Call Transfer			
• 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	SOFTSWITCH	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			
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Chapter 4: Scheduled Access Restrictions

Contents

This section contains information on the following topics:

Feature description

The Scheduled Access Restrictions (SAR) feature allows a customer to define Trunk Group Access Restrictions (TGAR), Class of Service (COS) restrictions, and Network Class of Service (NCOS) restrictions for different hours and days (typically off-hours and off-days). These TGAR, COS, and NCOS restrictions comprise SAR groups. Each customer may define up to 1000 SAR groups, and one of these groups can be assigned to each customer station or route. Up to eight time periods can be defined for each SAR group, and different restrictions may be applied to each time period.

SAR can be overridden on a single call basis for a station or route by using an authorization code or forced charge account. By using the SAR Disable (SADS), SAR Enable (SAEN), SAR Lock (SALK), or SAR Unlock (SAUN) Flexible Feature Codes (FFC), these restrictions can be changed on a more permanent basis.

SADS returns the telephone/route to its normal restriction state. SAEN cancels SADS, returning the telephone to its SAR state. SALK will occur automatically at a predefined period of time or when the Lock command is dialed by the user. Lock restrictions remain in effect until an SAUN or SADS command is entered. The SALK command can be used on a customer basis or SAR group basis, depending on the authcode used.

Typically, the Flexible Feature Codes can be used to do the following:

- extend off-hour restrictions for weekends or holidays (SALK)
- return to the schedule of access restrictions (SAUN)
- extend normal restrictions into the off-hour period for after hour services (SADS)
- cancel this after hour service (SAEN)
- cause off-hour restrictions to start immediately (SALK followed by SAEN)
- disallow any calls on an attendant console (SALK on SAR group containing the attendant(s))

Customer attendants that are included in SAR groups are placed in Position Busy when an off-hour or off-day period goes into effect. The restricted attendant can only release existing calls or dial the SAR Flexible Feature Codes. New calls cannot be made. Incoming calls will be directed to any other attendants that are not included in SAR groups and that are not in Position Busy.

If the system is placed in Night Service by an attendant, or the system is automatically placed in Night Service because all attendants are in the Position Busy state, incoming calls are routed to the Night DN. Going into Night Service will automatically place attendants who belong to a SAR group into a SAR Locked and Enabled state. These attendants can only release existing calls or dial the SAR Flexible Feature Codes; they cannot make new calls when restricted by SAR.

Operating parameters

The definition of authorization codes for SAR decreases the number of authorization codes available for non-SAR use.

SAR does not apply to Direct Inward System Access (DISA) DNs. DISA can be used to manually modify the SAR schedule using an FFC authorization code.

Telephones and trunks assigned to SAR groups have their Class of Service (COS), Trunk Group Access Restriction (TGAR) and Network Class of Service (NCOS) defined by the SAR schedule of their SAR group.

During the periods that a SAR or SAR lock is in effect, the Controlled Class of Service (CCOS) for the station or trunk is overridden.

If a Facility Restriction Level (FRL) is changed in order to be associated with a different NFCR tree, the NCOS using that FRL is affected. Also, different FRLs, and therefore different New Flexible Code Restriction (NFCR) trees, are used at different times according to the NCOS assigned to the SAR group.

Feature interactions

Access Restrictions

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

Automatic Redial

The Scheduled Access Restrictions (SAR) on Automatic Redial (ARDL) redialed calls are set when the call is initiated. If restrictions are changed later, the prior restrictions still apply.

Attendant Clearing during Night Service

Attendant Clearing during Night Service should be equipped with Scheduled Access Restrictions (SAR) due to the fact that, when Night Service is in effect, the only operations that may be performed from attendant consoles which are members of a SAR group are:

- release any existing calls, or
- dial one of the following SAR FFCs:
    - Scheduled Access Disable (SADS)
    - Scheduled Access Enable (SAEN)
    - Scheduled Access Lock (SALK)
    - Scheduled Access Unlock (SAUN)

Authorization Code Security Enhancement

Authorization Codes can be used to override SAR restrictions. In addition, Authorization Codes are defined for the specific use of SAR FFCs.

Basic Alternate Route Selection

If SAR is equipped when Basic Alternate Route Selection (BARS) is configured, a NCOS value between 0 and 99 must be defined for each time period.

Call Detail Recording

If configured, Call Detail Recording (CDR) A-type records are printed for SAR Flexible Feature Codes functions.

Charge Account, Forced

Forced Charge Account (FCA) can be used to override Scheduled Access Restrictions (SAR) on a per-call basis, provided the current Class of Service (COS) of the telephone or trunk is CUN, TLD, or CTD. The current COS is the COS in force according to the SAR schedule. If an Authorization Code that sets the COS to CUN, TLD, or CTD is dialed before the FCA, the call is allowed. FCA sets the COS to UNR and the Network COS (NCOS) to the NCOS defined in LD 15, provided that FCA is enabled on both a customer and telephone/trunk basis.

Class of Service

Telephones defined in LD 10 and 11, and trunks defined in LD 14 which are assigned a SARG number, have their Class of Service defined by the SAR schedule of their SAR group.

Controlled Class of Service

During normal hours, Controlled Class of Service (CCOS) restrictions override normal telephone restrictions. During off-hour periods or times when a Scheduled Access Restrictions (SAR) Lock is in effect, however, Scheduled Access Restrictions apply. When the Lock or off-hour period ends, CCOS restrictions continue to apply until they are removed or SAR becomes effective again. Whether a CCOS controller or electronic lock is used to activate CCOS, there is no indication to the user when Scheduled Access Restrictions are in effect, overriding CCOS restrictions. A telephone defined in LD 10 or 11 or a trunk defined in LD 14, which is assigned a SAR group number, has its Class of Service defined by the SAR schedule of its SAR group.

Coordinated Dialing Plan

If SAR is equipped when Coordinated Dialing Plan (CDP) is configured, a NCOS value between 0 and 99 must be defined for each time period.

Direct Inward System Access

Direct Inward System Access (DISA) numbers are not assigned to SAR groups and therefore are not affected by SAR schedules.

DISA can be used to manually modify the SAR schedule, provided that the correct FFC and Authorization Code are dialed.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The SAR feature overrides Electronic Lock.

Multi-Tenant Service

If a SAR is assigned to a tenant, any telephone belonging to the tenant will follow this SAR schedule unless the telephone belongs to a SAR group. The telephone's Scheduled Access Restrictions override any SAR assigned to the tenant.

Network Alternate Route Selection

If SAR is equipped when Network Alternate Route Selection (NARS) is configured, a NCOS value between 0 and 99 must be defined for each time period.

Network Class of Service

When a Network Class of Service (NCOS) is changed, it may be necessary to alter the NCOS values defined for each SAR group in LD 88. The NCOS value, which defines the facility restriction level and hence the NFCR trees, is used as defined by the SAR schedule. Also, different FRLs, and hence different NFCR trees, are used at different times according to the NCOS assigned to the SARG.

New Flexible Code Restriction

If a Facility Restriction Level (FRL) is changed to be associated with a different NFCR tree, any NCOS which uses that FRL is affected. In turn, the NCOS assigned to a SAR group may also be affected.

Office Data Administration System

Office Data Administration System (ODAS) can be used to indicate that telephones have been assigned to a SAR group. ODAS must be equipped in order to print members of a SAR group in LD 81.

Position Busy with Call on Hold

If an attendant in a SAR group has a call on hold, the attendant is not placed in Position Busy when the group enters an off-hour period.

Speed Call, Network Speed Call

The System Speed Call and Network Speed Call features ignore the COS and TGAR access restrictions in a SAR schedule, using the COS and NCOS defined in the speed call list.

Trunk Group Access Restriction

SAR does not alter the Trunk Group Access Restriction defined per route.

Feature packaging

Scheduled Access Restrictions requires the following packages:

- Scheduled Access Restrictions (SAR) package 162
- Call Detail Recording (CDR) package 4 for CDR functionality
- Network Authorization Code (NAUT) package 63
- Multi-Tenant Service (TENS) package 86 for Multi-Tenant functionality

- Flexible Feature Codes (FFC) package 139 and Basic Authorization Codes (BAUT) package 25 for the manual modification of schedules
- Network Class of Service (NCOS) package 32 to make NCOS restrictions effective
- Charge Account for CDR (CHG) package 23, Charge Account/Authorization Code (CAB) package 24, and Forced Charge Account (FCA) package 52 for additional billing information

Feature implementation

Task summary list

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

1. Table 1: LD 88 on page 70

   Configure Scheduled Access Restrictions data block.
2. Table 2: LD 88 on page 71

   Print the status of the tenant SAR group
3. Table 3: LD 88 on page 71

   Configure the Authcode data block not to automatically generate Authcodes.
4. Table 4: LD 88 on page 72

   Define SAR entries in the Authcode entries data block.
5. Table 5: LD 10 on page 72

   Assign individual analog (500/2500-type) telephones to the selected SAR group in response to the SGRP prompt.
6. Table 6: LD 11 on page 73

   Assign individual proprietary telephones to the selected SAR group in response to the SGRP prompt.
7. Table 6: LD 11 on page 73

   Assign individual attendant consoles to the selected SAR group in response to the SGRP prompt.
8. Table 7: LD 12 on page 73

   Assign individual trunk route to the selected SAR group in response to the SGRP prompt.

9. Table 8: LD 16 on page 73

   Define Flexible Feature Codes for the SAR disable, SAR enable, SAR lock, and SAR unlock functions.

10. Table 9: LD 57 on page 74

    Assign a SARG for each tenant by responding to the TEN prompt with the tenant number and the SGRP prompt with the number of the SAR group to be assigned to the tenant.

11. Table 10: LD 93 on page 74

    Reset current time of day to activate the feature and SAR schedule.

Table 1: LD 88

Prompt	Response	Description
REQ	NEW CHG	Create or change existing data block.
TYPE	SAR	Scheduled Access Restrictions.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15). Prompt will not appear to a user with an LAO password.
SGRP	0-999	SAR group number.
SCDR	(NO) YES	(Do not) activate CDR for the SAR FFC commands.
OFFP	1-8	Off-hour period number. Off-hour periods may overlap; the period that starts first has priority until that off-hour period is over.
	<CR>	Go to ICR prompt.
- STAR hh mm	hh mm	Start time. The current start time (hours and minutes) is printed individually after the prompt. Respond with the new start time.
	X	Remove value and return to OFFP prompt.
- STOP hh mm	hh mm	Stop time. The current stop time (hours and minutes) is printed individually after the prompt. Respond with the new stop time.
	X	Remove value and return to OFFP prompt.
- DAYS	d ... d	Respond with a new set of days to be used. Maximum of seven entries in the range of 1-7. Day 1 = Sunday, Day 2 = Monday, etc.

Prompt	Response	Description
- COS	(UNR) CTD CUN FR1 FR2 FRE SRE TLD	Off-hour period Class of Service. Unrestricted Conditionally Toll-Denied Conditionally Unrestricted Fully Restricted Class1 Fully Restricted Class 2 Fully Restricted Semi-restricted Toll Denied
- TGAR	(0)-15	Trunk Group Access Restriction.
- NCOS	0-99	Network Class of Service.
- ICR	(NO) YES	Incoming Calls are Restricted.
LOCK	(1)-8	The LOCK prompt is used to indicate which off-hour period is to be used as the LOCK period. The default is Period 1.

Table 2: LD 88

Prompt	Response	Description
REQ	PRT	Print.
TYPE	SAR	Scheduled Access Restrictions.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password.
TEN	1-511	Tenant number.
SGRP	0-999	Prompted only if no tenant number is entered.

If the system is in an off-hour or locked period when a print command is issued, an asterisk appears following the restrictions being used. If lock is in effect, an additional asterisk appears following the lock prompt.

Table 3: LD 88

Prompt	Response	Description
REQ	NEW	New.
TYPE	AUB	Authcode data block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15).
ALEN	1-14	Number of digits in authcodes.
ACDR	YES NO	Activate CDR for authcodes (there is no default response).
RANR		RAN route number for "Authcode Last" prompt (NAUT)
	0-511	Range for Large System and CS 1000E system.

Prompt	Response	Description
CLAS	(0)-115	Classcode value assigned to authcode.
AUTO	NO	Do not automatically generate Authcodes. Prompted when NAUT package 63 is equipped and REQ = NEW. The Authcode length must be a minimum of four digits.

Table 4: LD 88

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	AUT	Authcode entries data block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15).
CODE	xxx..x	Authcode (1-14 digits).
SARC	YES NO	Allow or disallow Authcode to be used as the SAR authorization code.
- SERV		SAR service functions for SARC (the SERV prompt appears if SARC = YES)
	(END) ENA (LKD) LKA (DSD) DSA (UND) UNA	Enable (Denied) Allowed. Lock (Denied) Allowed. Disable (Denied) Allowed. Unlock (Denied) Allowed Up to four entries can be made at once.
- SRGP	0-999 ALL	Number of SAR group to be defined or changed. Change all SAR groups.
CLAS	(0)-115	Class code value assigned to authcode. Cycle continues with CODE. When type = AUT, enter X to configure the authcode as an exempt code. When this data is printed, the month the authcode was deactivated is output. The default is 0 when adding authcode entries.
	X	Exempt authcode.

Table 5: LD 10

Prompt	Response	Description
REQ:	New CHG	Add, or Change.
TYPE:	500	Analog (500/2500-type) telephone.
TN		Terminal number

Prompt	Response	Description
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Table 6: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add, or 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.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Table 7: LD 12

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Attendant consoles do not follow the SAR restrictions defined by the SGRP, but they can be locked by using SAR FFCs

Table 8: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Table 9: LD 57

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
...		
CODE	aaaa	Specific Flexible Feature Code Type. To change a specific Flexible Feature Code, enter the associated mnemonic then carriage return <CR>. The mnemonic will then be prompted and the Flexible Feature Code can be entered. The Flexible Feature Code may be up to four digits or up to seven digits if DNXP package 150 is equipped.
- SADS	xxxx	SAR Disable code.
- SAEN	xxxx	SAR Enable code.
- SALK	xxxx	SAR Lock code.
- SAUN	xxxx	SAR Unlock code.

Table 10: LD 93

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TGEN	Tenant SAR data block.
CUST	xx	Customer number, as defined in LD 15
...		
TEN	1-511	Tenant number.
...		
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.

Feature operation

Modification of SAR Restrictions

SAR restrictions can be modified on a per call basis by using an Authorization Code, if the Basic Authorization Code (BAUT) package 25 is equipped.

Also, if the Authorization Code and Flexible Feature Codes packages are equipped, the off-hour periods can be shortened or extended by using the four SAR FFCs.

The Authorization Code feature can be used to allow a user to override a Scheduled Access Restriction on a single-call basis by dialing an Authorization Code (Authcode). Each Authcode is assigned a Class of Service, Trunk Group Access Restriction, and a Network Class of Service. The restrictions associated with the dialed Authcode apply to the call being made. Thus, by using an Authcode, any facility to which access is allowed depending on the restrictions associated with an Authcode, can be accessed by dialing the telephone, even though the telephone may normally be denied access.

Single-Call Modification

The Scheduled Access Restrictions feature does not modify using Authcodes to allow calls to be made on restricted sets. Dial either SPRE + 6 or the AUTH FFC plus the Authcode associated with the desired restrictions. Once dial tone is returned, indicating a valid code, the call may be dialed as normal.

Off-Hour Period Modification

The SADS, SAEN, SALK, and SAUN FFCs defined in LD 57 can be used to modify off-hour period restrictions, by simply dialing the FFC plus an appropriate Authcode. The Authcode determines if the requested function is allowed and whether the action is to take place on a SAR group or a customer basis. An FFC plus and Authcode for a specific SARG is only accepted from a station within that group, or from a station within a tenant which uses that SAR group.

Entering a Flexible Feature Code plus an Authcode results in the following:

- SALK + Authcode = extend off-hour restrictions for weekends or holidays
- SAUN + Authcode = return to the schedule of access restrictions
- SADS + Authcode = extend normal restrictions into the off-hour period for after hour services

- SAEN + Authcode = cancel this after hour service
- SALK followed by SAEN + Authcode = cause off-hour restrictions to start immediately, and
- SALK on SAR group containing the attendant(s) + Authcode = disallow any calls on an attendant console.

Chapter 5: Scheduled Electronic Lock

Contents

This section contains information on the following topics:

Feature description

Scheduled Electronic Lock (SELK) enhances the Electronic Lock feature.

The Scheduled Electronic Lock feature automatically locks telephone sets at predetermined times. These times are defined in the Scheduled Access Restrictions (SAR) database (LD 88). SAR group numbers are also defined in LD 88. A maximum of eight scheduled lock times can be assigned to each group.

Each telephone that requires Scheduled Electronic Lock functionality must be assigned to a SAR group in LD 10 or 11, and must have the Scheduled Electronic Lock Allowed (SLKA) Class of Service assigned.

In order to override the Scheduled Electronic Lock feature, the user must use the existing Electronic Lock feature. The user enters the Electronic Lock Deactivated (ELKD) Flexible Feature Code (FFC). The telephone remains unlocked until the user dials the Electronic Lock Activated (ELKA) FFC. If the user does not dial the ELKA FFC, the system automatically locks the telephone at the next scheduled lock time. For the telephone to be unlocked again, the user must dial the ELKD FFC to unlock the telephone. The telephone does not automatically unlock.

A special dial tone, defined in LD 56, notifies the user that the telephone is in a locked state.

Scheduled Electronic Lock Example

The Scheduled Electronic Lock is scheduled for 18:00, 24:00, 02:00. At 22:00, an employee who is working overtime needs to use their telephone. That person enters the ELKD FFC on the telephone to unlock it. At 24:00, the telephone automatically locks, if it has not already been locked by the user. To use the telephone again, the employee must unlock it. At 02:00, the next scheduled lock time, the telephone locks once more. The Scheduled Electronic Lock feature remains in effect until the employee unlocks the telephone by dialing the ELKD FFC.

Operating parameters

The Scheduled Electronic Lock feature supports analog (500/2500-type) and digital telephones on Remote Office.

The Scheduled Electronic Lock feature does not support ACD sets, trunks or PC Attendant.

If a telephone does not support the SAR and Electronic Lock features (for example, ACD sets), then it will not support the Scheduled Electronic Lock feature.

If the Class of Service (CLS) is configured to Scheduled Electronic Lock Deactivated (SKLD) in LDs 10 and 11, the existing Electronic Lock and SAR feature functionality apply.

When the Scheduled Electronic Lock feature is active, it does not take the Controlled Class of Service (CCOS) restriction from LD 15. Configuration is done in LD 88.

When a telephone is unlocked, CCOS restrictions (if active) override normal telephone restrictions. When the Scheduled Electronic Lock feature is active, the Scheduled Access Restrictions override the CCOS restrictions.

If the system is busy the Scheduled Electronic Lock feature could be slightly delayed. In this case, it is possible that a user could still dial an external number after the beginning of a scheduled lock time.

Feature interactions

Automatic Call Distribution

The Scheduled Electronic Lock feature does not support Automatic Call Distribution (ACD) sets, as CCOS does not support ACD sets.

Direct Inward System Access

Direct Inward System Access (DISA) numbers are not assigned to Scheduled Access Restrictions groups, so they are not affected by the SELK feature.

Electronic Lock Network Wide / Electronic Lock on Private Lines

The SELK feature supports Electronic Lock Network Wide / Electronic Lock on Private Lines. However, a scheduled lock is not supported over a network. Scheduled Electronic Lock must be configured and administered locally. Like SELK, these features obtain their restrictions from Scheduled Access Restrictions.

Message Intercept

When SELK locks a telephone, Message Intercept (MINT) provides a different dial tone or announcement while the telephone is locked.

Multi Tenant Service

If Scheduled Access Restrictions are applied to a tenant, the telephones in that tenant group follow the Scheduled Access Restrictions (unless the telephone belongs to a different SAR group).

Scheduled Access Restrictions

The Scheduled Access Restrictions (SAR) Permanent Disable, Active Lock, and Lock Disable FFCs have precedence over SELK.

Feature packaging

Scheduled Electronic Lock requires the following packages:

- Controlled Class of Service (CCOS) package 81
- Flexible Feature Code (FFC) package 139

- Scheduled Access Restrictions (SAR) package 162
- Message Intercept (MINT) package 163, if the Message Intercept function is required

Feature implementation

Task summary list

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

1. Table 11: LD 88 on page 80

   Configure the Scheduled Access Restrictions data block.
2. Table 12: LD 10 on page 81

   Configure the Scheduled Electronic Lock feature on an analog (500/2500-type) telephone.
3. Table 13: LD 11 on page 82

   Configure the Scheduled Electronic Lock feature on a digital telephone on Remote Office.
4. Table 14: LD 57 on page 82

   Define Flexible Feature Codes for Scheduled Electronic Lock.

Table 11: LD 88

Prompt	Response	Description
REQ	NEW CHG	Create data block. Change existing data block.
TYPE	SAR	Scheduled Access Restrictions.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (same password as defined for DISA on a per customer basis in LD 15). This prompt does not appear to a user with an LAO password.
SGRP	0-999	Scheduled Access Restrictions group number.
OFFP	1-8	Off-hour period number. Off-hour periods can overlap; the period that starts first has priority until that off-hour period is finished. All the prompts shown up to the ICR prompt repeat until you enter <CR>.
	<CR>	Go to the ICR prompt.

Prompt	Response	Description
- STAR hh mm	hh mm	Start time. The current start time (hours and minutes) is printed individually after the prompt. Respond with the new start time.
- STOP hh mm	hh mm	Stop time. The current stop time (hours and minutes) is printed individually after the prompt. Respond with the new stop time.
- DAYS	d ... d	Respond with a new set of days to be used. Maximum of seven entries in the range of 1-7. Day 1 = Sunday, Day 2 = Monday, etc.
- COS	(UNR) CTD CUN FR1 FR2 FRE SRE TLD	Off-hour period Class of Service. Unrestricted Conditionally Toll-Denied Conditionally Unrestricted Fully Restricted Class1 Fully Restricted Class 2 Fully Restricted Semi-restricted Toll Denied
- TGAR	0-(1)-15	Trunk Group Access Restriction.
- NCOS	0-99	Network Class of Service.
ICR	(NO) YES	Incoming Calls are Restricted.
LOCK	(1)-8	Indicates off-hour period to be used as the LOCK period. Default is Period 1.

Table 12: LD 10

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	500	Analog (500/2500-type) 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.
DES	x...x	ODAS Station Designator.
CUST	xx	Customer number, as defined in LD 15
...		
SCPW	xxxx	Station Control Password. SCPL must be configured in LD 15.
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.
...		
CLS	CCSA	Controlled Class of Service Allowed. CCSD = Controlled Class of Service Denied (default).

Prompt	Response	Description
	SLKA	Scheduled Electronic Lock Allowed. SLKD = Scheduled Electronic Lock Denied (default).

Table 13: LD 11

Prompt	Response	Description
REQ:	NEW	Add new 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.
DES	x...x	ODAS Station Designator.
CUST	xx	Customer number, as defined in LD 15
...		
SCPW	xxxx	Station Control Password. SCPL must be configured in LD 15.
SGRP	(0)-999	Scheduled Access Restriction group number. Must have group defined in LD 88.
...		
CLS	CCSA	Controlled Class of Service Allowed. CCSD = Controlled Class of Service Denied (default).
	SLKA	Scheduled Electronic Lock Allowed. SLKD = Scheduled Electronic Lock Denied (default).

Table 14: LD 57

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FFC	Flexible Feature Code.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Provide FFC confirmation tone.
CODE	ELKA	New/change Electronic Lock Activate FFC.
ELKA	xxxx	Enter the new or changed Electronic Lock Activate FFC.
CODE	ELKD	New/change Electronic Lock Deactivate FFC.
ELKD	xxxx	Enter the new or changed Electronic Lock Deactivate FFC. ELKD must be different than ELKA.

Feature operation

During the period that a telephone is locked, the user must enter the ELKD FFC to unlock the telephone. The telephone remains unlocked until either the user dials the ELKA FFC to manually lock the telephone or the next scheduled lock occurs.

Chapter 6: Secrecy Enhancement

Contents

This section contains information on the following topics:

Feature description

This feature allows a warning tone to be applied to a three-way connection involving the source, destination and attendant if Warning Tone Allowed (WTA) Class of Service is available on both the source and destination sides. If the warning tone is denied on either the source or destination, these parties are automatically split. This applies to all calls handled by the attendant instead of only incoming network calls and attendant recalls with the original secrecy feature. The warning tone is always applied to a three-way connection.

There will be no connection established through the console with more than two parties, excluding the attendant, unless all parties have WTA Class of Service.

This feature also prevents any intelligible crosstalk on an attendant-held call or if the source (SRC) or destination (DEST) party is excluded.

Operating parameters

A connection is not established through the console if one of the parties, excluding the attendant, has warning tone denied Class of Service.

Feature interactions

AC15 Recall: Timed Reminder Recall

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

Attendant Break-In with Secrecy

The source and destination parties cannot be joined together on the attendants conference bridge if BKIS is active. This is consistent with the existing Break-In feature.

Attendant Recall

When the attendant answers a recall, the attendant is automatically connected to the destination party and the source party is excluded.

Semi-Automatic Camp-On

Secrecy and Enhanced Secrecy apply to Semi-automatic Camp-On recalls, with splitting taking place when the attendant answers the recall.

Secrecy

All functionalities of the Secrecy feature apply to the Secrecy Enhancement feature.

Slow Answer Recall Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Source Included when Attendant Dials

Source Included when Attendant Dials takes precedence over Secrecy and Enhanced Secrecy.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 15: LD 15 - Select secrecy enhancement for customer:

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	FTR	Features and options Data Block
...		
OPT	(SYD) SYA EHS	Secrecy allowed Enhanced Secrecy allowed Secrecy denied

Feature operation

No specific operating procedures are required to use this feature.

Chapter 7: Secretarial Filtering

Contents

This section contains information on the following topics:

Feature description

Secretarial Filtering is an application of Call Forward All Calls. It allows you to forward all calls to a second telephone. The user at the second telephone answers the forwarded calls and can choose to transfer the call back to you.

In the following example, a manager having a secondary Directory Number (DN) of 2644 forwards all calls arriving at that DN to a secretary's secondary DN 2744. Any call placed to DN 2644 is forwarded to the secretary at DN 2744. The secretary answers the call, decides that the manager should take the call, and transfers it back to DN 2643 (the prime DN). In this example, the manager receives only the calls originated or transferred by the secretary.

Operating parameters

Only the Directory Number (DN) designated as the Call Forward number can originate or transfer calls to the originally dialed DN.

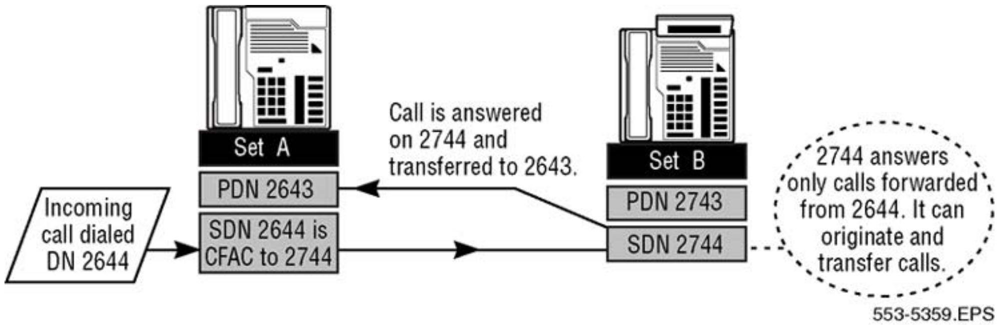


Figure 1: Secretarial Filtering example

All Single Appearance DNs on the forwarded telephone are forwarded to the target DN.

A Multiple Appearance DN on the forwarded telephone is forwarded only if it is a Prime DN. A Multiple Appearance DN that is not the Prime DN rings at all appearances, including the forwarded telephone.

Feature interactions

Call Forward/Hunt Override Via Flexible Feature Code

The Secretarial Filtering feature is overridden by the Call Forward/Hunt Override Via FFC feature, but there are no changes to the feature itself.

Network Intercom

In a Secretarial filtering scenario, the secretary BFS lamp will also reflect that the boss telephone is busy if the boss is on a Hot Type I call.

Phantom Terminal Numbers (TNs)

If a Phantom TN is call forwarded to an existing telephone, and that telephone is used to call a DN on the Phantom TN, the call receives DCFW treatment.

Feature packaging

Secretarial Filtering is included in base system software. It is provided with Call Forward All Calls.

Feature implementation

This feature is enabled when Call Forward All Calls is enabled.

Feature operation

See the feature operation in the Call Forward All Calls module in this document.

Chapter 8: Seizure Acknowledgment

Contents

This section contains information on the following topics:

Feature description

Outgoing Ear and Mouth (E&M) Direct Inward Dialing (DID) or Direct Outward Dialing (DOD) trunks with an immediate start arrangement may require a seizure acknowledgment signal be received after a trunk seizure. This signal is an off-hook message. If the signal is not received within one second of the seizure, the trunk is software busied for three seconds, then dropped. The outgoing call then attempts to seize the next trunk in the sequence to complete the call. If the signal is received, the call is processed normally.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Seizure Acknowledgment feature described above.

The Seizure Acknowledgment feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Table 16: LD 16 - Set Seizure Acknowledgement for a trunk route.

Prompt	Response	Description
REQ	aaaa	Request (aaaa = CHG, END, LCHG, NEW, OR OUT)
TYPE	RDB	Type of data block = RDB (Route data block)
CUST	xx	Customer number, as defined in LD 15
...		
ACKW	(NO) YES	Seizure acknowledgment signal (is not) is expected after seizure of this DID/DOD trunk.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 9: Selectable Conferee Display and Disconnect

Contents

This section contains information on the following topics:

Feature description

The Selectable Conferee Display and Disconnect (SCDD) feature expands existing Conference Display functionality and provides Meridian Modular (Aries) telephone users and IP Phone users with the capability to selectively drop any party that has been added to a conference. This feature provides both Meridian Modular telephones and IP Phones involved in a conference with the following two enhancements:

- Conference Count Display
- Selectable Conferee Disconnect

Important:

The Selectable Conferee Display and Disconnect feature only applies to Meridian Modular telephones or IP Phones equipped with a display screen and involved in a conference with at least three conferees.

Conference Count Display

When on a conference call with Conference Count Display activated, Meridian Modular telephones and IP Phones show the following information on their display screens:

- the elapsed time
- the total number of parties currently active on the call (includes all conferees in the conference, regardless of the type of phone used: Meridian Modular, IP Phone, or other)

Whenever a conferee is added to or disconnects from the conference, the Conference Count Display updates to reflect the change.

To activate the Conference Count Display, configure the Class of Service to Conferee Display Count Allowed (CDCA). The display screen then shows only the elapsed time, not the number of conferees in attendance.

The Conference Count Display divides into three fields. You must configure at least one of these fields for Conference Count Display to work. You can configure the fields from the Customer Data Block. The fields are as follows:

- The Total Conferees Count display field (CNFFIELD) shows the total number of parties involved in a conference (total internal conferees plus total external conferees). The default mnemonic for this field on the display screen is CONF.
- The Total Internal Conferees Count display field (INTFIELD) shows the total number of conferees internal to the system. This includes analog (500/2500-type) telephones, Meridian 1 proprietary telephones, IP Phones, attendant consoles, and service trunks (such as Paging, Music, and Recorded Announcement) within the system. The default mnemonic for this field on the display screen is I.
- The Total External Conferees Count display field (EXTFIELD) shows the total number of conferees external to the system. This includes trunks connected to the system that can be configured on internal or external routes. The default mnemonic for this field on the display screen is E.

You can modify the mnemonics for each of the above fields to accommodate different languages or to save output time. To make modifications, you must define the CNF_NAME, INT_NAME and EXT_NAME prompts in the Customer Data Block. You can use a mnemonic of one to four characters in length for each of the three fields.

Important:

Avaya recommends that you take the real time impact into consideration when modifying the mnemonics for the three display fields. Since the displays on both Meridian Modular telephones and IP Phones receive each character individually (including spaces), configuring the maximum number of characters in the field headings (four characters each) increases the refresh time for each telephone involved in the conference. Taking real time impact into consideration is especially important in conferences involving a large number of parties.

Figure 2: Meridian Modular telephone display with all three display fields enabled in LD 15 on page 97 offers an example of a Meridian Modular telephone in a conference with five parties: three internal conferees and two external conferees. With the Class of Service configured to Conference Display Count Allowed (CDCA), the conference count information shows up on the left part of the sample display. The conference count information breaks down as follows:

- enabling CNFIELD in the Customer Data Block shows as CONF 5 in the sample display, where the total number of conferees in attendance equals 5.
- enabling INTFILED in the Customer Data Block shows as I 3 in the sample display, where the total number of internal conferees in attendance equals 3.
- enabling EXTFIELD in the Customer Data Block shows as E 2 in the sample display, where the total number of external conferees in attendance equals 2.

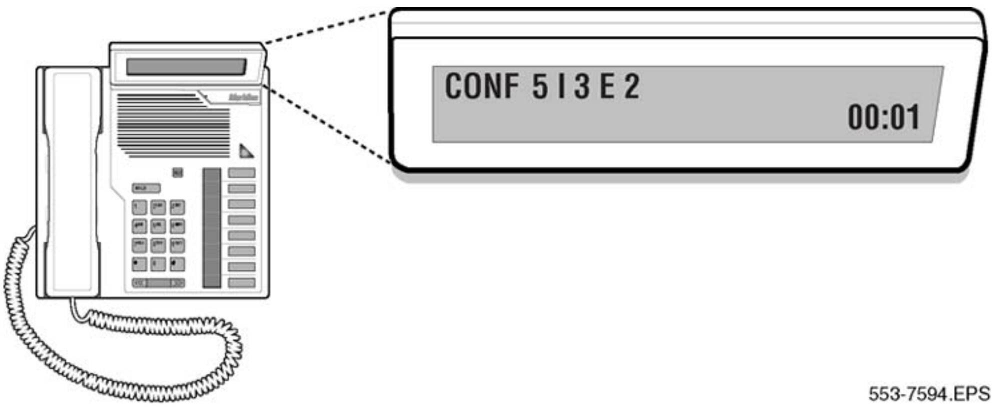


Figure 2: Meridian Modular telephone display with all three display fields enabled in LD 15

From the Customer Data Block, you can combine the three Conference Count Display fields to configure up to eight possible Conference Count Display formats. Figure 3: Possible display formats of the active Conference Count Display on page 98 offers an example of a conference hosting a total of six conferees: four internal parties and two external parties.

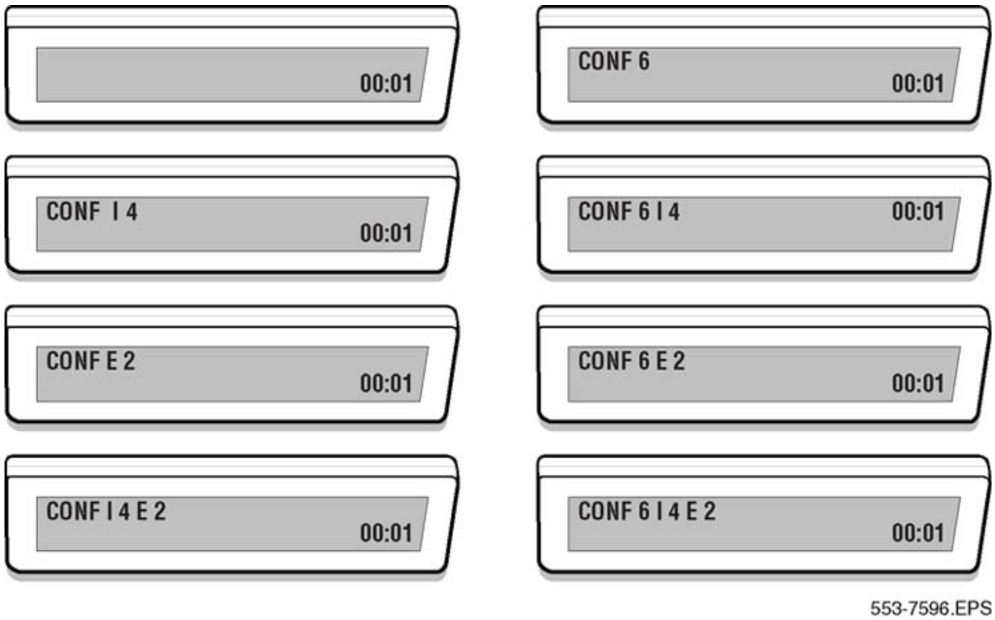


Figure 3: Possible display formats of the active Conference Count Display

The eight possible formats from the preceding graphic are organized as follows:

- The left column shows display formats with CNFIELD disabled (no total number appears next to CONF).
- The right column shows display formats with CNFIELD enabled (the total number 6 appears next to CONF).
- The first row shows display formats with both INTFIELD and EXTFIELD disabled.
- The second row shows display formats with INTFIELD enabled (the total number appears as I 4) and EXTFIELD disabled.
- The third row shows display formats with INTFIELD disabled and EXTFIELD enabled (the total number appears as E 2).
- The fourth row shows display formats with both INTFIELD and EXTFIELD enabled (totals appear as I 4 and E 2).

Important:

To display the Total Conferees Count display field name (CNF_NAME), you must configure at least one of the CNFFIELD, INTFIELD, or EXTFIELD prompts to YES in the Customer Data Block.

Each display field on the screen of either a Meridian Modular telephone or an IP Phone shows a maximum conferee count of six. If the total number of conferees exceeds six, then the Conference Count Display field shows as 6+. Figure 4: Meridian Modular telephone display in a conference with more than six conferees on page 99shows a Meridian Modular telephone display in a conference consisting of more than six internal parties (I 6+) and more than six external parties (E 6+). Therefore, the Total Conferees Count also exceeds six (CONF 6+).

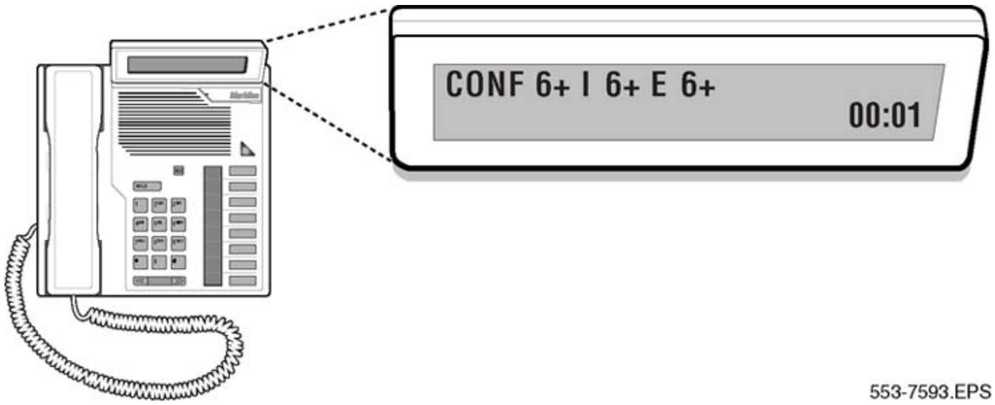


Figure 4: Meridian Modular telephone display in a conference with more than six conferees

Figure 5: Example of a Conference Scenario involving both internal and external parties on page 100 demonstrates an active conference with five internal telephones and three external trunks. The display screens of the Meridian Modular telephones show the Conference Count Display information.

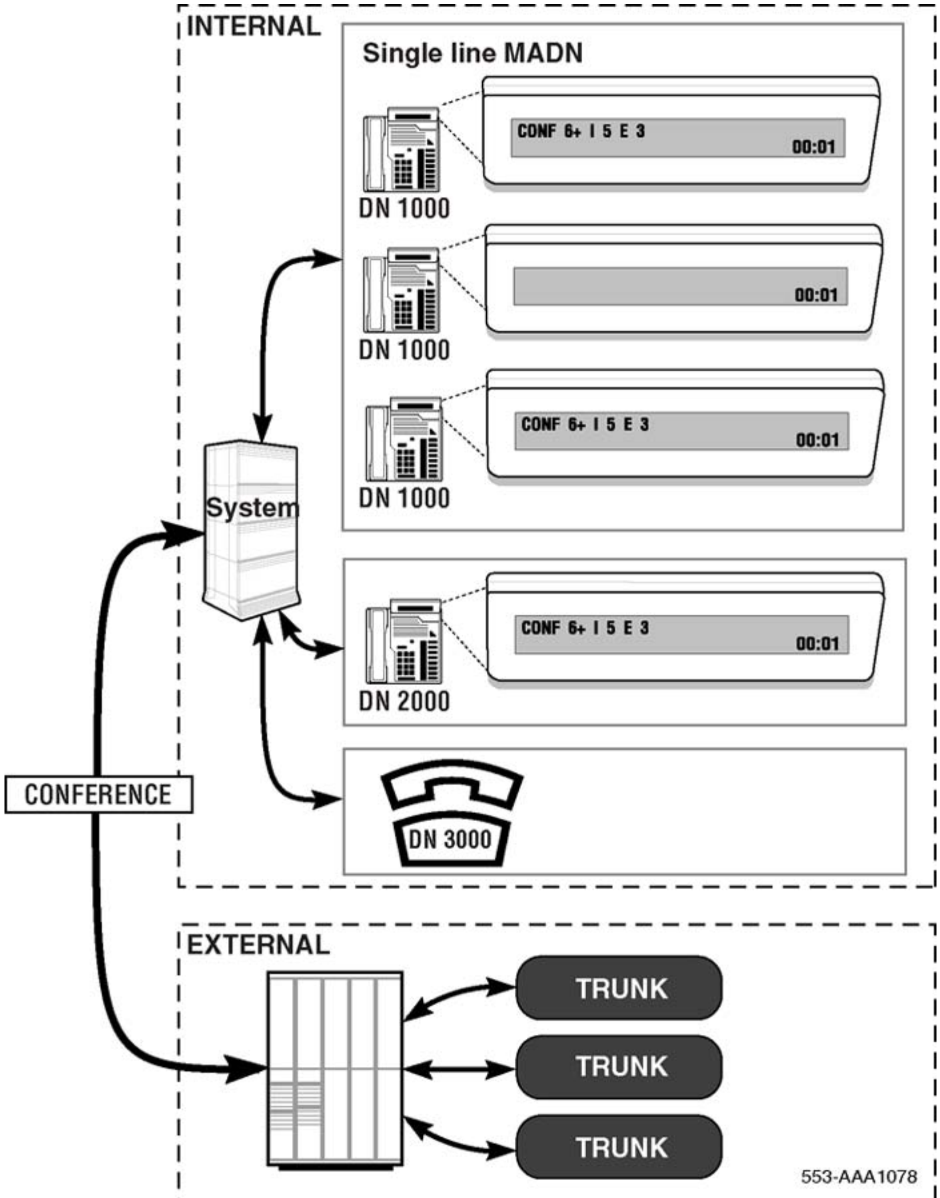


Figure 5: Example of a Conference Scenario involving both internal and external parties

In the preceding example:

- Three Meridian Modular telephones show as DN1000, a single line Multiple Appearance Directory Number (MADN).
- Of the three telephones on DN1000, two entered the conference through Privacy Override.
- One of the telephones on DN1000 has its Class of Service configured to CDCD in LD 11 (its display shows only elapsed time), while all other Meridian Modular telephones in the scenario have their Class of Service configured to CDCA allowed.

- The active conference also includes DN2000 (a Meridian Modular telephone) and DN 3000 (an analog 500/2500 type telephone).
- All three display fields are enabled in LD 15.

Selectable Conferee Disconnect

With Selectable Conferee Disconnect, a Meridian Modular telephone or IP Phone user scrolls through a list of active conferees, using a Conferee Selectable Display (CSD) key. You can configure the CSD key at the telephone level.

To activate Selectable Conferee Disconnect, press the CSD key during an active conference. The screen of the CSD key user can display every conferee involved in the conference (except CSD key users themselves).

With the CSD key in use, the display format of each conferee follows the existing simple two-party call display. The display shows the name and extension number of the conferee. If the conferee is on a trunk, the display shows the trunk group access code and the trunk member number.

As the key user, after you activate the CSD key you can then selectively disconnect any displayed conferee from the conference by pressing the active call key (the key on which the conference is established). The CSD key user can also cancel the Selectable Conferee Disconnect operation by pressing the Release key.

When you first press the CSD key, the last conferee to join the active conference displays first. Press the CSD key again, and the telephone displays another conferee from the list (in no particular order). Each time you press the key, the display scrolls forward through the list of all conferees. If you scroll past the party that you want to disconnect, keep pressing the CSD key until the list cycles through to the party you are looking for.

When you activate the CSD key, the CSD key lamp lights up. If, however, the displayed conferee cannot be disconnected (as in the case of an attendant console), then the Key lamp flashes. The display screen remains unchanged and continues to show the same conferee on the display.

The following figures show the display screen of telephone A, a Meridian Modular telephone, as it displays and disconnects a selected conferee:

- Figure 6: Display before CSD key activation on page 102
- Figure 7: Display with CSD key activated and a conferee selected on page 102
- Figure 6: Display before CSD key activation on page 102
- Figure 8: Display after disconnecting conferee on page 102

In these examples, telephone A operates under the following parameters:

- Class of Service set to CDCA in LD 11
- CSD key configured in LD 11
- CNFFIELD key configured in LD 15 (the telephone displays only the Total Conferees Count)

Figure 6: Display before CSD key activation on page 102 shows a total of six conferees involved in an active conference.

Figure 6: Display before CSD key activation

In Figure 7: Display with CSD key activated and a conferee selected on page 102, telephone A presses the CSD key and scrolls to conferee, Terry Smith.

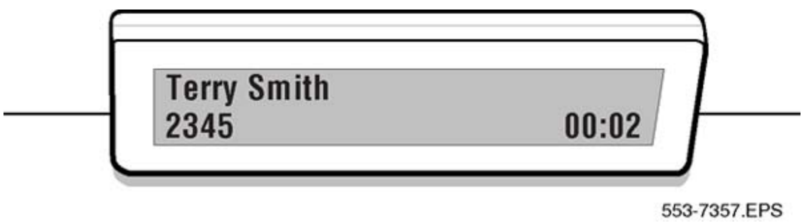


Figure 7: Display with CSD key activated and a conferee selected

In Figure 8: Display after disconnecting conferee on page 102, telephone A presses the active call key to disconnect Terry Smith. Telephone A's display screen is updated to show the new total conferee count status after the conferee is disconnected from the active conference. The Conference Count Displays of the other Meridian Modular telephones or IP Phones involved in the conference are also updated. After the disconnection of Terry Smith, a total of five conferees remain active in the established conference.

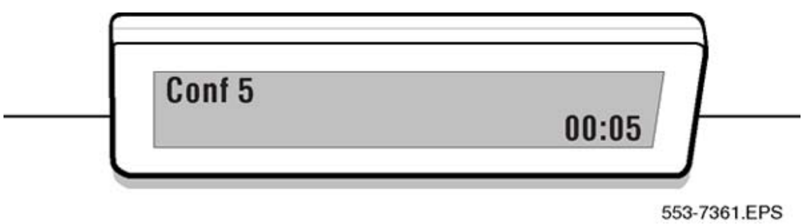


Figure 8: Display after disconnecting conferee

Operating parameters

The Selectable Conferee Display and Disconnect feature only applies to Meridian Modular telephones or IP Phones equipped with a display screen and involved in an active conference with at least three conferees in attendance.

Meridian Modular telephones include M2008, M2016, M2616, M2216ACD1, and M2216ACD2 telephones. IP Phones include the Avaya 2033 IP Conference Phone, 2001 IP Phone, 2002 IP Phone, 2004 IP Phone, Avaya 2007 IP Deskphones, Avaya 2050 IP Softphone, Avaya Mobile Voice Client 2050 for Personal Digital Assistants (PDAs), WLAN Handset 2210, and WLAN Handset 2211.

When conferees disconnect from an active conference, leaving only two parties left, the conference usually converts to a simple two-party call. Situations where the two parties remain in conference mode include:

- one or both parties are attendant consoles
- both conferees are on mixed telephones using the same DN.

The Selectable Conferee Display and Disconnect feature does not apply to two-party conferences.

Simple call display for the last two remaining parties in a conference is as per existing operation.

The methods used to add a conferee to a conference do not affect Selectable Conferee Display and Disconnect. These methods include: 3-party and 6-party Conference, Override, Attendant Barge-In, Attendant Break-In, and Bridging.

The two Selectable Conferee Display and Disconnect sub-features, Conference Count Display and Selectable Conferee Disconnect, have independent functions and operations.

Conference Count Display

To activate Conferee Count Display in LD 11, configure the Class of Service to Conferee Display Count Allowed (CDCA). You must enable at least one of the three Conference Count Display field options in the Customer Data Block.

To configure the Class of Service to Conferee Display Count Allowed (CDCA) in LD 11, you must first configure either the Automatic Digit Display (ADD) or the Delay Display (DDS).

To configure a display screen to show the elapsed time only (existing functionality):

- Set all three Conference Count Display field options to NO in LD 15.
- Set the Class of Service to Conferee Display Count Denied (CDCD) in LD 11.

Selectable Conferee Disconnect

To activate Selectable Conferee Disconnect, you must define a Conferee Selectable Display (CSD) key in LD 11. Before defining the CSD key, however, you must configure the Class of Service in LD 11 to either Automatic Digit Display (ADD) or Delay Display (DDS).

You can configure only one CSD key per Meridian Modular telephone or IP Phone.

The CSD user's telephone displays each conferee (internal and external) according to existing simple two-party call display. Defining the CSD key does not change any of the features that supply or display conferee data. These features include: Call Party Name Display, Calling Party Privacy, Dialed Number Identification Service, Digit Display, Display of Calling Party Denied, and ISDN Calling Line Identification.

For conferees on a trunk, the display may not show the specific terminating telephone. For conferences with several trunks in attendance, Avaya recommends that you keep a record of each party that party joins the conference and on which trunk.

When a conferee uses the CSD key, the displays (if there are any) on the other conferee telephones do not change. Only the CSD key user can view the list of conferees.

After you press the CSD key, you can only use the following keys: the active call key, the Release (RLS) key, and the CSD key. All other input is ignored.

If the CSD key user goes on-hook with the CSD key activated, then the key user is disconnected from the call, not the displayed conferee.

The Selectable Conferee Disconnect feature does not support the use of confirmation tones to indicate when a conferee is disconnected from the active conference.

If the system initializes or sysloads during an active conference, the conference is torn down as per existing functionality. If the CSD key is active when the system initializes or sysloads, then the key operation is canceled.

Usually, the active conferee list displays the last party to join the conference first. However, if the last party to join uses the CSD key, then the active conferee list displays in no particular order. This happens for two reasons: 1) only the last party to join is displayed with any priority; and 2) CSD key users are never displayed. Once the last party becomes a CSD key user, all priority disappears from the list, which then shows in no particular order. Likewise, if the last party is disconnected from the conference, the list then shows in no particular order.

You can use the CSD key to disconnect a conferee from the conference at any time.

Pressing a key or feature key replaces the active conference display is replaced. The Conference Count Display does not restore until a conferee either joins or leaves the conference, thereby updating the conferee count. If the conferee is placed on hold and then restored, the Conference Count Display appears once again.

Feature interactions

Attendant

With the CSD key activated, the active conference can display the attendant console as a conferee. You cannot use the CSD key to disconnect an attendant console from the conference. Only the attendant console can release itself from a conference call.

Any attempt to disconnect the attendant console using the CSD key causes the CSD key lamp to flash. To stop the lamp from flashing, press the Release key (cancelling the CSD key operation) or press the CSD key again (scrolling to the next conferee).

Attendant Barge-In

When an attendant barges into a conference, the action separates all conferees into either the destination (DEST) side or the Source (SRC) side. Conferees connected through the trunk that is being verified move to the DEST side (not including the attendant). All other conferees, including the attendant, move to the SRC side. However, all parties can communicate with each other.

Once the conference is established on either the SRC or the DEST side, you can use the CSD key. However, you cannot disconnect an attendant.

Attendant Break-In

To extend an incoming call to a busy destination DN, the attendant must first break into the active call. To break into an active call, the attendant presses the Break-In key. The attendant then aks the destination DN to disconnect from the active call. After the destination DN disconnects, the attendant can extend the incoming call to the destination DN.

If the attendant breaks into a simple two-party call, the call changes from a two-party call to a three-party conference (including the attendant). Once established on the conference, the Conference Count Display no longer shows on conferee telephones. Instead, both conferee telephones show the attendant information only.

When breaking into a conference call, the attendant effectively joins the conference. The Conference Count Display no longer shows on any conferee telephone. Instead, conferee telephones display attendant information only. After the destination DN disconnects from the conference (at the request of the attendant), the attendant can then extend the incoming call to the destination DN. If the remaining attendees continue the conference, then the Conference Count Display reappears on those attendee telephones.

The CSD key can be used during an established conference. However, the incoming call does not show as a conferee. The attendant does show as a conferee, but you cannot use the CSD key to disconnect an attendant.

Attendant Administration

You can modify Attendant Administration (AA) in LD 71 to print the CSD key. However, you cannot use AA to configure a CSD key.

Automatic Call Distribution

An Automatic Call Distribution (ACD) agent or supervisor can activate Conference and No Hold Conference. If your ACD telephone is a Meridian Modular telephone or IP Phone equipped with a display and a CSD key, then you can use the Selectable Conferee Display and Disconnect feature.

Agent Observe

Selectable Conferee Display and Disconnect does not change the functionality of the ACD Agent Observe feature. While in the observe mode:

- The conference does not include the ACD supervisor.
- The active conference count does not include the ACD supervisor.
- The ACD supervisor's telephone does not show the Conference Count Display.
- With the CSD key activated, the ACD supervisor does not show in the active conferees list.

ACD Agent Features

Avaya recommends that you do not assign the CSD key to an agent's telephone, because the CSD key can be used to disconnect a supervisor.

Alternate Call Answer

With ACAA set to YES in LD 23, an agent can place an active Individual DN (IDN) call on hold. With the IDN call on hold, the agent can then press the In-Calls key to return to the idle agent queue and take the next call. To establish a conference with the agent, the IDN call, and the ACD call, the agent must activate Call Join.

With Alternate Call Answer, after you establish an active conference with the ACD agent, an IDN call, and the ACD call, you can then apply the Selectable Conferee Display and Disconnect feature.

Agent and Supervisor Communication

To include the supervisor on a simple call with an ACD caller, the ACD agent presses the ACD supervisor (ASP) key. To answer, the supervisor presses the Agent (AGT) key. To finish the operation, the agent waits until the supervisor answers, then presses the ASP key.

When in a conference call with an ACD caller, the ACD agent cannot use the ASP key to add a supervisor to the conference.

With Agent and Supervisor Communication, after you establish an active conference established, you can then apply the Selectable Conferee Display and Disconnect feature.

Emergency Key

The ACD agent can use the Emergency Key (EMR) feature to conference an ACD supervisor and, optionally, activate a recording device for customer-defined emergencies or sensitive situations.

With the EMR key activated, the recording trunk is not considered a member of the conference. With the CSD key activated:

- The active conferee list does not include the recording trunk.
- The total number of conferees on the Conference Count Display does not include the recording trunk.

ACD Display Enhancement

With ACD Display Enhancement, you cannot press the Not Ready (NRD) key while using the Conference key. If you press the NRD key while in an active conference call, the conference call disconnects, the NRD lamp illuminates, and the 'NOT READY' screen appears.

You cannot use the NRD key while the CSD key remains activated.

ACD In-Calls Key

If you establish a conference on the ACD In-Calls key, you can then use the In-Calls key to drop a desired conferee. As the CSD key user scrolls through the active conferees list, the Position Identification (POS ID) of each ACD telephone involved in the conference is displayed.

Application Module Base

The Selectable Conferee Display and Disconnect feature uses existing messaging to disconnect a conferee. It uses the same messaging as when a conferee either goes on-hook or presses the Release (RLS) key to disconnect themselves from the conference.

Automatic Hold

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Automatic Hold feature. After you establish a conference on the active DN key, you can then apply the Selectable Conferee Display and Disconnect feature.

Basic Rate Interface

BRI telephones do not support the Selectable Conferee Display and Disconnect feature. However, with the CSD key activated, conferees using BRI telephones do show their information in the conference.

Bridging

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Bridging feature.

With the Bridging feature, the same DN can appear on up to eight single-line telephones. Any appearance of the Multiple Appearance Directory Number (MADN) can enter a call by going off-hook. When you use Bridging to establish a three-party conference, the conference includes only two active DNs. Conferee telephones display only the information for the other DN on the call, not the Conference Count Display information. However, since there are three conferees on the call, you can use the CSD key. Once the conference expands to include more than two active DNs, Conference Count Display then shows the conferee count on connected telephones.

Conference

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Conference feature, except for the new active conference display. Conference calls can include calls on the following key types: Single Call Arrangement DN (SCN, SCR), Multiple Call Arrangement DN (MCN, MCR), ACD In-Calls (ACD DN), Private Line Ringing and Non-ringing (PLN, PLR), Hotline (HOT), Call Waiting (CWT), Voice Call (VCC) and Dial Intercom (DIG).

Conference Control

The Selectable Conferee Display and Disconnect feature does not change the functionality of the Conference Control feature.

Digitone Receiver

The Selectable Conferee Display and Disconnect feature does not treat the Digitone Receiver (DTR) as a conferee when it appears on the conference loop, since it appears only temporarily to provide the tone service.

Display Key

While in a conference call, you can use the Display (DSP) key to obtain information. However, the Display key is blocked when the CSD key is active.

DNIS Across Call Modifications

When a CSD key user scrolls through the list of conferees during a DNIS call, DNIS information is displayed.

End-to-End Signaling

With the CSD key activate, the Selectable Conferee Display and Disconnect feature does not block End-to-End Signaling (EES) or dialing digits.

Group Call

The Selectable Conferee Display and Disconnect feature only applies to the originator of a Group Call involving three or more active parties. The active conference display does not show until a redisplay of the Group Call originator's screen is needed.

Hold

With the Selectable Conferee Display and Disconnect feature, when display-equipped Meridian Modular telephones or IP Phones involved in a conference show the Conference Count Display. If a Meridian Modular telephone or IP Phone places the conference on hold (by pressing the Hold key), then the active DN key lamp flashes, and the display clears for the duration of the Hold operation. The Conference Count Display is restored upon completion of the Hold operation. To take the call off hold, press the active DN key.

Meridian Link

The Selectable Conferee Display and Disconnect feature uses existing messages sent over the Meridian Link to provide the Conference Count Display and the Selectable Conferee Disconnect functionality.

No Hold Conference

The Selectable Conferee Display and Disconnect feature does not change the No Hold Conference (NHC) functionality. The Selectable Conferee Display and Disconnect feature applies to conferences created by No Hold Conference.

Avaya Integrated Conference Bridge

The Selectable Conferee Display and Disconnect feature does not change the functionality of Avaya Integrated Conference Bridge.

Override

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Override (OVR) feature. The Conference Count Display does not show for an Override conference, because the Override display shows instead. The CSD key, however, can be used to disconnect conferees in an Override conference.

Priority Override

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Priority Override (POVR) feature. The Conference Count Display does not show for a POVR conference, because the Priority Override display shows instead. The CSD key can, however, be used to disconnect conferees involved in a POVR conference.

Privacy

The Selectable Conferee Display and Disconnect feature does not affect the operation of the Privacy feature. With Privacy enabled, only one appearance of a single line MADN can participate in a conference call. The conference counts include this MADN appearance.

Privacy Override

The Selectable Conferee Display and Disconnect feature does not change the operation of the Privacy Override (POA) feature.

A Meridian 1 proprietary telephone or IP Phone with a Class of Service configured to Privacy Override Allowed (POA) can bridge into an established call on a single line MADN. When you create a conference with three parties through Privacy Override, the conference call includes only two active DNs. In this case, conferee telephones display only the information for the other DN on the call, not the Conference Count Display information. However, since there are three conferees on the call, you can use the CSD key. Once the conference expands to include more than two active DNs, the Conference Count Display then shows the conferee count on connected telephones. The Conference Count Display totals include conferees added to the conference through POA. You can apply the CSD key on established conferences.

Tone and Digit Switch

The Selectable Conferee Display and Disconnect feature does not treat the Tone and Digit Switch (TDS) as a conferee when it appears on the conference loop, because it appears only temporarily to provide tone service.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

- Table 17: LD 15 on page 112

  Configure the Conference Count Display Format for the customer.
- Table 18: LD 11 on page 113

  Set the Conferee Display Count Allowed (CDCA) Class of Service for Meridian Modular telephones and IP Phones.
- Table 19: LD 11 on page 113

  Configure a Conferee Selectable Disconnect (CSD) key for Meridian Modular telephones and IP Phones.

Table 17: LD 15

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		
CONF_DSP	YES	Change Conference Count Display configurations. NO = Do not change Conference Count Display configurations (default). To prompt for further conference display options, CONF_DSP must be YES.
- CNFFIELD	(NO) YES	Total Conferees Count display field (disabled) enabled.

Prompt	Response	Description
- CNF_NAME	(CONF) aaaa	Total Conferees Count display field name. Enter 1-4 alphanumeric characters to replace the existing name. The Total Conferees Count display field name is displayed when any of the CNFFIELD, INTFIELD, or EXTFIELD prompts are YES.
- INTFIELD	(NO) YES	Total Internal Conferees Count display field (disabled) enabled.
--INT_NAME	(I) aaaa	Total Internal Conferees Count display field name. Enter 1 to 4 alphanumeric characters to replace the existing name.
- EXTFIELD	(NO) YES	Total External Conferees Count display field (disabled) enabled.
--EXT_NAME	(E) aaaa	Total External Conferees Count display field name. Enter 1 to 4 alphanumeric characters to replace existing name.

Table 18: 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	ADD DDS	Automatic Digit Display. Delay Display. With CLS = DDS, the display is activated after the call is answered. CLS must be either ADD or DDS prior to setting CLS = CDCA or CDCD.
	(CDCA)	Conferee Display Count Allowed (default) CDCD = Conferee Display Count Denied. CDCD option telephones a blank display screen during a conference call.

Table 19: LD 11

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

Prompt	Response	Description
...		
CLS	ADD DDS	Automatic Digit Display. Delay Display. With CLS = DDS, the display is activated after the call is answered. CLS must be either ADD or DDS prior to configuring a CSD key.
KEY	xx CSD	Conferee Selectable Display key. To remove the CSD key, set the KEY prompt to xx NUL, thereby disabling Selectable Conferee Disconnect.

Feature operation

Viewing the list of active conferees

To view the list of active conferees, press the Conferee Selectable Display (CSD) key. To view the next conferee in the list, press the CSD key again. The CSD key lamp illuminates. The displays on the other Meridian Modular telephones or IP Phones involved in the conference do not change.

To cancel the Selectable Conferee Disconnect operation, press the Release key. This prevents the disconnection of any of the conferees. The CSD key lamp remains unlit. If enabled, the Conference Count Display returns. The original conference call remains active throughout this operation.

Disconnecting one conferee

To disconnect a conferee using the CSD key:

- Press the CSD key repeatedly until the conferee that you want to disconnect appears on the screen. The CSD key lamp illuminates. The displays on other Meridian Modular telephones or IP Phones do not change.
- Press the active call key (the key on which you established the active conference). The CSD key lamp remains unlit. If enabled, the Conference Count Display then shows the revised total count of conferees. The original conference call remains active throughout this operation.

Disconnecting more than one conferee

To disconnect more than one conferee, follow the steps for disconnecting a single conferee. Each conferee must be disconnected separately.

Important:

When two CSD key users want to drop different conferees (but not each other), each CSD key user can initiate the Selectable Conferee Disconnect operation and disconnect the selected conferee. If enabled, the Conference Count Displays on Meridian Modular telephones or IP Phones are updated after each successful Selectable Conferee Disconnect operation.

Disconnecting the same conferee

Two Meridian Modular telephones or IP Phones (telephone A and telephone B), both equipped with a CSD key, want to disconnect the same conferee. The telephone that presses the active call key first succeeds in disconnecting the conferee. If telephone A presses the active call key first, its Conference Count Display updates with the revised total count of conferees. The Conference Count Display of all other Meridian Modular telephones or IP Phones, with the exception of telephone B, are also updated. Telephone B's Conference Count Display does not update until the user presses either the active call key or (to end the operation) the Release key.

Verifying that a conferee has been disconnected

To verify that a conferee has been disconnected:

- Use the CSD key to view the list of conferees. Note whether or not the disconnected conferee appears in the list.
- Check the CSD key lamp. An unlit lamp indicates that the Conferee Selectable Disconnect operation is complete.
- Check that the display screen shows the revised total count of conferees on the Conference Count Display.
- If the disconnected conferee leaves only two remaining parties in the conference, the conferee switches to a simple call situation, and the displays update accordingly.

Canceling the Selectable Conferee Disconnect operation

To cancel Selectable Conferee Disconnect at any time during the operation, press the Release key. This prevents any conferee from being disconnected. The CSD lamp remains unlit and the Conference Count Display (if enabled) returns. The original conference call remains active throughout this operation.

Disconnecting from an active conference

To disconnect yourself from an active conference, press the Release key, or go on-hook. As long as a supervised conference situation remains, the original conference call remains active.

Chapter 10: Selectable Directory Number Size

Contents

This section contains information on the following topics:

Feature description

The Selectable Directory Number Size feature allows a user to define the number of digits that must be received on a Direct Inward Dialing (DID) route before the end of dialing (EOD) is reached. If the required number of digits is not received when the EOD timer expires, a TRK137 message is sent to print and the trunk is locked out.

The DN size can be specified from one to seven digits, or as zero which will not consider the number of digits dialed in the sequence.

Operating parameters

The Public Exchange/Central Office must be equipped to handle the special signaling requirements associated with the Seizure Acknowledgment feature described above.

The Seizure Acknowledgment feature is not available on 1.5 Mbps digital trunks or Japanese Digital Multiplex Interface (DMI) trunks.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

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

Feature implementation

Table 20: LD 16 - Set limits for the Selectable Directory Number Size feature.

Prompt	Response	Description
REQ	aaaa	Request (aaaa= CHG, END, LCHG, NEW, or OUT
TYPE	RDB	Type of data block = RDB (Route data block)
CUST	xx	Customer number, as defined in LD 15
...		
DNSZ	(0)-7	Number of digits expected on DID routed; 0 indicates no fixed number.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 11: Semi-Automatic Camp-On

Contents

This section contains information on the following topics:

Feature description

This feature allows a Camp-On call to recall to the attendant instead of ringing the called party when the called party becomes available. The called party can originate calls but cannot receive any other calls. Other incoming calls to this DN will receive a busy indication. If the called party originates another call when the attendant attempts to present the Camp-On call, the attendant receives busy tone and can initiate Camp On again or release the call.

When an attendant extends a call to a desired party that is busy, the attendant can activate Semi-automatic Camp-On by pressing the Semi-automatic Camp-On (SACP) key. This causes the call to be camped-on to the desired party, and recalled to the attendant when the desired party becomes idle, rather than rung through to the desired party.

Recall to Same Attendant must be allowed, otherwise the recall is routed to the first available attendant. The attendant display shows the calling-party DN and the party to which the call is camped-on. If the attendant, or all attendants in a multiple-console environment, are busy then the recall is placed in the attendant queue.

Meanwhile, incoming calls to the desired party receive busy treatment. The desired party, however, is still able to make calls. After receiving the recall, the attendant can ring the desired party by pressing the SACP key. The attendant may release the call while it is ringing, or hold the call until it is answered. If the desired party has made another call while the attendant tries to present the recall, the attendant may Camp-On the recall to the desired party by pressing the SACP key.

Operating parameters

The same operating parameters apply as for Camp-On.

Semi-automatic Camp-On is mutually exclusive with the Call Waiting feature. Thus, attendant consoles configured with Semi-automatic Camp-On will not work if Call Waiting has been defined.

Semi-automatic Camp-On can be configured for individual or all Camp-On occurrences.

Semi-automatic Camp-On is not available with Network Attendant Service. If the attendant tries to apply Semi-automatic Camp-On to a station at a remote node, the SACP lamp flashes to indicate that Semi-automatic Camp-On is not allowed. The attendant has to press the SACP key again to deactivate the feature, and be allowed to activate it under normal operation.

Semi-automatic Camp-On is not supported during Night Service or Enhanced Night Service. Calls that were camped-on by Semi-automatic Camp-On during normal hours ring through to the desired party, when idle, and do not recall to the attendant.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature uses the SACP key to activate a blocking attempt, but the Attendant Blocking of DN feature is only valid on the source side of the attendant console. The Semi-automatic Camp-on feature is only valid on the destination side of the attendant console.

To have the Attendant Blocking of DN feature available and not the Semi-automatic Camp-on feature, a new response to the SACP prompt has been introduced in LD 15. Prompt SACP = NO means the Semi-automatic Camp-on feature is not available even if the SACP package is equipped and an SACP key exists on the attendant console. To have the Semi-automatic Camp-on feature available the SACP prompt must be answered with SNGL or ALL which have the same meanings as before.

Attendant Break-In

The attendant can Break-In to an established call and apply Semi-automatic Camp-On to the desired party. The attendant may press the SACP key before or after the Break-In.

Call Forward/Hunt Override Via Flexible Feature Code

Semi-Automatic Camp-On can be used even if the Call Forward/Hunt Override Via FFC feature is activated. When encountering a busy telephone, it is possible to activate SACP, if it is applicable.

Incoming calls during recall

During Semi-automatic Camp-On, when the desired party becomes idle and the camp-on is recalled to the attendant, the desired party appears busy to incoming calls. The DN of the desired party is displayed as busy on the Busy Lamp/Enhanced Busy Lamp display.

Periodic Camp-On Tone

Periodic Camp-On Tone stops when the camped-on call is recalled to the attendant.

Secrecy and Enhanced Secrecy

Secrecy and Enhanced Secrecy apply to Semi-automatic Camp-On recalls, with splitting taking place when the attendant answers the recall.

Source Included when Attendant Dials

The source remains included while the attendant dials the destination.

Feature packaging

This feature requires Semi-automatic Camp-On (SACP) package 181.

Feature implementation

Task summary list

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

1. Table 21: LD 15 on page 122

   Configure Semi-automatic Camp-On for the customer.
2. Table 22: LD 12 on page 122

   Configure an SACP key on the attendant console.

Table 21: LD 15

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	ATT_DATA	Attendant console options.
...		
RTSA	RSAA	Recall To Same Attendant Allowed.
SACP	(NO) SNGL ALL	Semi-automatic Camp-On. Semi-automatic Camp-On not allowed. Enable Semi-automatic Camp-On on a per-call basis. Enable Semi-automatic Camp-On for all occurrences. SACP keys must be defined on all attendant consoles which are to make use of the feature.

Table 22: LD 12

Prompt	Response	Description
REQ	CHG	Change
TYPE	2250	Attendant console type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
KEY	xx SACP	Key Number, Semi-automatic Camp-On

Feature operation

When an attendant extends a call to a desired party who is busy, the attendant can activate Semi-automatic Camp-On as follows:

- Press the SACP on the attendant console. The call is camped on the desired party.
- The display on the attendant console shows the calling party's DN, and the party to which the call is camped on (the desired party).
- The desired party becomes idle. The call is recalled to the attendant.
- To ring the desired party after receiving the recall, press the SACP on the attendant console again.

Recall timing on Camp-On calls

When any station extends an external call, recall timing will be initiated if the call is camped on to a busy station.

The recall timing will start from the moment that the extending station releases the call. The value of the recall timer is configured by the prompt RTIM in the Customer Data Block (LD 15).

At the recall, the camped on call will be routed to the attendant. If the attendant is in Night Service, Night treatment is given; if NAS routing is active, the call will be routed according to the NAS configuration.

Standalone case

When the recall to the attendant occurs, the Camp-On is canceled. If the attendant is busy during the recall, the recall will be queued.

Network case

When the recall occurs and the attendant has answered the recall, the call will still be camped on to the desired party. If during the recall the attendant is busy, the recall will be queued.

Chapter 12: Series Call

Contents

This section contains information on the following topics:

Feature description

The Series Call feature causes a source call (either an attendant-answered incoming call, or an attendant-originated trunk call), that has been extended to an internal destination party, to be recalled to the attendant when the destination party hangs up. The attendant can then extend the source call to another destination party. This feature enables a caller to talk to more than one party without having to disconnect and call again (Recall to Same Attendant must be allowed, otherwise the recall is routed to the first available attendant). This process can be repeated for as many destinations as requested by the caller.

A Series Call is canceled if one of the following occurs:

- the attendant presses the Series Call (SECL) key while the associated lamp is lit
- the attendant extends the source to a trunk while the SECL lamp is lit
- the attendant enters Night Service after extending the call and prior to receiving the recall
- the destination is call forwarded to a trunk, or
- the source disconnects.

Operating parameters

This feature only applies when the destination party is internal. If the attendant dials a DN that is not internal, the SECL key will flash to indicate that the feature cannot be invoked.

The source can only be extended to an internal party.

Feature interactions

Attendant Position Busy

If the attendant activates Position Busy while a Series Call is active, the recall occurs to the next available attendant.

Call Detail Recording

With Call Detail Recording, a start record is generated when a source Periodic Pulse Metering call is answered and marked as a Series Call by the attendant, and an end record is generated when the attendant releases the call. No intermediate records are generated.

Night Service

If the attendant extends a Series Call and goes into Night Service before it recalls to the attendant, the call recalls to the night DN and Series Call treatment is canceled.

Timed Reminder Recall

With Timed Reminder Recall, if the attendant extends a Series Call during Camp-on, Call Waiting, or ringing, the SECL lamp goes dark.

Feature packaging

This feature requires Series Call (SECL) package 191.

Feature implementation

Task summary list

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

1. Table 23: LD 12 on page 127

   Configure Series Call for each attendant console.
2. Table 24: LD 15 on page 127

   Configure Recall to Same Attendant for the customer.

Table 23: LD 12

Prompt	Response	Description
REQ	CHG	Change
TYPE	2250	Attendant console type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
KEY	xx SECL	Key Number, Series Call

Table 24: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.

Prompt	Response	Description
...		
- RTSA	(RSAD) RSAA	Recall (Denied) Allowed to Same Attendant.

Feature operation

The attendant designates the source call as a Series Call by pressing the Series Call (SECL) key. The SECL key may be pressed by the attendant while dialing, talking to the destination party, or while a call is ringing. The associated key lamp remains lit until the Series Call is canceled. If the attendant tries to extend a call to an external station, the SECL lamp flashes. The attendant has to press the SECL key to cancel the Series Call, and extend the call as a standard call extension.

Chapter 13: Set-Based Administration Enhancements

The Set-Based Administration Enhancements feature provides Set-Based Administration for all system types.

For more information about the Set-based Administration Enhancements feature, see *Avaya Set-Based Administration, NN43001-603*.

Chapter 14: Short Buzz for Digital Telephones

Contents

This section contains information on the following topics:

Feature description

When a call is presented to a digital telephone that is off-hook, a buzz tone is given. The duration of this secondary buzz is shortened from two seconds to an average of 0.8 seconds, with a minimum length of 0.5 seconds and a maximum length of one second.

Operating parameters

Short Buzz for digital sets does not change the buzz tone given to Automatic Call Distribution (ACD) telephones on the In-calls key.

Feature interactions

Directory Number Delayed Ringing

If a telephone is defined with Directory Number Delayed Ringing (DNDR) delay and there is an incoming call to another SCN/MCN DN key on the same telephone, buzzing (or short buzzing) is applied after the DNDR delay timer expires.

Group Call

The special three-second buzz for Group Call is not affected by this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 15: Single Appearance Directory Number

Contents

This section contains information on the following topics:

Feature description

A Single Appearance Directory Number (SADN) can be assigned to any type of telephone.

Operating parameters

A Single Appearance Directory Number (SADN) is a DN that appears only once within a customer group.

Feature interactions

Directory Number Expansion

The DN can have up to seven digits if the Directory Number Expansion package is equipped.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 25: LD 10 on page 134

   Assign a Directory Number.

2. Table 26: LD 11 on page 135

   Assign Single Appearance Directory Number keys.

Table 25: LD 10

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Analog (500/2500-type) telephone.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
...		
DN	x...x	Directory Number. Up to four digits; up to seven digits with DNXP package 150.

Table 26: 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.
KEY	xx SCN yyy...y	Add a single-call non-ringing DN key, where: xx = key number, and yyy...y = DN.
	xx SCR yyy...y	Add a single-call ringing DN key, where: xx = key number, and yyy...y = DN.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 16: Single-digit Access to Hotel Services

Contents

This section contains information on the following topics:

Feature description

In hospitality applications, it is desirable for room phones to have single-digit access to hotel services and a multiple-digit access to room phones.

The Single-digit Access to Hotel Services feature allows a customer to define a pause timer, called a second-digit timer, between the first and second dialed digits, and allows two speed-call entries to be defined for a station group. The first speed-call entry is used for normal pretranslation. The second speed-call list is used when the second digit timer times out (that is, when time out occurs after the first digit is dialed, with the first digit in the first speed-call list being translated).

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

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

Dependency:

- Pretranslation (PXLT) package 92

Feature implementation

Task summary list

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

1. Table 27: LD 15 on page 138

   Enable Single-digit Access to Hotel Services for each customer.
2. Table 28: LD 18 on page 139

   Define the Translation tables required by this feature.

Table 27: LD 15

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and Options.
...		
- OPT	(SDDE) SDAL	(Deny) allow Single Digit Access.

Table 28: LD 18

Prompt	Response	Description
REQ	CHG	Change
TYPE	PRE	Pretranslation calling group data block
CUST	xx	Customer number, as defined in LD 15
XLAT	xxx yyyy	Calling group number to translation Speed Call list number correlation. Format if International Supplementary Features (SUPP) package 131 is not equipped • • xxx = Pretranslation group number, 0-254 • xxx = Group 0 is used for trunks • xxx = Group 1 is used for attendant consoles. • xxx = Groups 2-254 can be used for other calling groups. • yyyy = List number to be used for Pretranslation, 0-8191. 8191 is used to remove the group from Pretranslation.
		Pretranslation group number. Format if international Supplementary Features (SUPP) package 131 is equipped • • xxx = Group 0 is used for trunks • xxx = Group 1 is used for attendant consoles. • xxx = Groups 2-254 can be used for other calling groups.
...		
- SDA	0-8190	Single-digit Access Speed Call List number

Feature operation

In the example that follows, if a room guest dials the digit 7, the guest's call is immediately terminated at DN 4300, the front desk. If the guest had dialed the digit 2, then after the second digit timer times out, the guest's call is terminated at DN 4002, laundry. If the guest enters three more digits (xxx) before the second digit time-out, the appropriate room number (2xxx) is rung.

Speed Call CodeDN Designation

0	Operator (00)
1	Room Service (4001)
2	Laundry (4002)
3	Concierge (4100)
4	Restaurant (4101)
5	Health Club (4200)
6	Maid (4201)
7	Front Desk (4300)
8	Toll Calls (88)
9	Local Calls (99)

First Entry Speed Call List (for normal pretranslation)

First Dialed DigitAction

1	Pass as 1
2	Pass as 2
3	Pass as 3
4	4101
5	4200
6	4201
7	4300
8	88
9	99
0	Pass as 0

Second Entry Speed Call List (for pretranslation after time out)

First Dialed DigitAction

1	4000
2	4002
3	4100
4	N/A
5	N/A
6	N/A

7	N/A
8	N/A
9	N/A
0	0017

Chapter 17: Slow Answer Recall Enhancement

Contents

This section contains information on the following topics:

Feature description

This enhancement to the Slow Answer Recall feature changes how the recall is treated once presented to the attendant console. This enhancement applies to Integrated Services Digital Network (ISDN) and standalone environments.

If an incoming call extended by the attendant to a telephone is not answered after a preprogrammed time period, it is recalled to the attendant console. The call type may be indicated by an Incoming Call Indicator (ICI) key programmed to flash for recalls. The target telephone will continue to ring after the call is presented to the attendant. The target telephone can answer the call before the attendant does, in which case the call is cleared from the attendant console and the incoming call and target telephone will be connected.

If the attendant answers the recall before the target telephone, a speech connection is established between the calling party on the source (SRC) side of the console. The target telephone continues to ring while still being connected to the destination (DEST) side of the console. This feature only affects the operation after the attendant has answered the recall.

When a Slow Answer Recall occurs, the call is placed in the attendant queue and appears on the console. The target telephone will continue to ring while the recall is queued and presented on the console, but is unanswered. When the attendant answers the recall, by pressing the appropriate Loop key or the Recall ICI key, the target telephone will be disconnected as soon as the attendant console answers the Slow Answer Recall.

In a ISDN environment, the feature works in a similar way regardless of the location of the called party (on the same node as the attendant or on a remote node), and if Network Attendant Service (NAS) routing is involved in the call or not.

Call Waiting Recalls and Camp-on Recalls

This enhancement adds Call Waiting Recall and Camp-on Recall functionality to Slow Answer Recall. This enhancement applies within standalone and networking environments.

Call Waiting Recall

Within a standalone environment, if an incoming call extended by the attendant or a telephone (equipped with the Multi-Party Operations feature) to a busy station (equipped with Call Waiting) is not answered within a customer-defined period of time, it is recalled to the attendant. The recall is presented to the attendant or placed in the attendant queue.

Camp-on Recall

An incoming call is extended by the attendant or a telephone (equipped with the Station Camp-on feature) to a busy station that is not equipped with Call Waiting. The attendant or telephone Camp-on the call to the target telephone. If the call is not answered within a customer-defined period of time, it is recalled to the attendant. The call is presented to the attendant or placed in the attendant queue. Until the attendant answers the call, the call remains camped-on to the target telephone and can still be answered. If the attendant answers the recall by pressing the appropriate loop key or the Recall key, the target telephone is disconnected and can no longer answer the call. The target telephone must be redialed to extend the call.

Within a network environment, the Call Waiting Recall and Camp-on Waiting Recall enhancements must be configured at a node. Both the Call Waiting Recall and Camp-on Waiting Recall enhancements operate in the same way as in the stand-alone case. The location of the calling and called party and the attendant have no affect on the call processing.

Network Attendant Service (NAS) is not required, but it may be applied at a node. In this case, NAS takes precedence over the Call Waiting Recall and Camp-on Waiting Recall Enhancements, in that the target telephone is disconnected from the call due to time-out and not to the attendant pressing the loop key or Recall key.

Operating parameters

The same as for Slow Answer Recall.

Feature interactions

Attendant Recall with Splitting, Multi-Party Operations , Secrecy Enhancement

The Call Waiting Recall and Camp-on Waiting Recall enhancements take precedence over Attendant Recall Splitting (ATS), Secrecy (SYA), Enhanced Secrecy (EHS), and Multiple Party Operations.

Call Waiting Recall, Camp-on Waiting Recall

The Call Waiting Recall and Camp-on Waiting Recall enhancements are compatible with Station Camp-on (STCA).

A forced Camp-on override recall occurs to the attendant. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are equipped, the destination is automatically disconnected when the attendant answers. If the Call Waiting Recall and Camp-on Waiting Recall enhancements are not equipped, and the attendant answers the recall at the same time that the destination answers, a conference is established between the attendant, source, and destination.

Intercept Computer Dial from Directory

If the attendant extends an SRC party to a DEST party on the local node, but slow answer recall occurs since the DEST does not answer, it is possible to dial a new DN from the ICP (the DEST is disconnected when the attendant answers).

Feature packaging

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

Feature implementation

Table 29: LD 15 - Enable Slow Answer Recall Enhancement for the customer.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and Options data block
...		
- OPT	(SLD) SLA	Slow Answer Recall Enhancement (denied) allowed.

Feature operation

Slow Answer Recall Enhancement

When a Slow Answer Recall occurs the call is placed in the attendant queue and appears on the console. The target telephone will continue to ring while the recall is queued and presented on the console but unanswered. When the attendant answers the recall, by pressing the appropriate Loop key or the Recall ICI key, the target telephone will be disconnected as soon as the attendant console answers the Slow Answer Recall.

Call Waiting Recall

Until the attendant answers the call, it remains waiting on the target telephone, and can still be answered. If the attendant answers the recall by pressing the appropriate Loop key or the Recall key, the target telephone is disconnected and can no longer answer the call – the target telephone will have to be redialed to extend the call.

Camp-on Recall

Until the attendant answers the call, the call remains camped-on to the target telephone, and can still be answered. If the attendant answers the recall by pressing the appropriate Loop key or the Recall key, the target telephone is disconnected and can no longer answer the call; the target telephone will have to be redialed to extend the call.

Chapter 18: Slow Answer Recall for Transferred External Trunks

Contents

This section contains information on the following topics:

Feature description

This feature allows an external call to be transferred to a ringing telephone anywhere within an Integrated Services Digital Network (ISDN) network. The transferred call may be incoming or outgoing, supervised or unsupervised. If the call is not answered within a customer-defined period of time, it is routed to the local attendant as a slow answer recall.

Within a standalone environment, this capability is provided by the Multi-Party Operation feature.

An external call is a call originated by the Public Switched Telephone Network (PSTN). This includes calls originating on a Central Office (CO), Foreign Exchange (FEX), Direct Inward Dialing (DID), or Wide Area Telephone Service (WATS) trunk on a local or remote node, and calls from the PSTN to an ISDN node using Network Attendant Service (NAS) signaling protocol over an ISDN TIE trunk.

Operating parameters

This feature applies only to systems using Meridian Customer Defined Networking (MCDN) signaling over ISDN Signaling Link (ISL)/ISDN TIE links.

All network nodes must be configured with Network Attendant Service (NAS).

Feature interactions

AC15 Recall: Transfer from Norstar

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

Attendant Recall

Slow Answer Recall Modification (SLAM) has an interaction after the attendant answers the recall. If SLAM is configured, the target telephone is disconnected after the attendant answers the recall. If SLAM is not configured, the target telephone rings until the attendant releases it.

Call Forward No Answer

If the ringing station to which the call has been transferred has Call Forward No Answer active, the call will be transferred to the call forward DN after the specified number of ring cycles.

ICP Network Screen Activation, Flexible DN Interactions

When an Intercept Computer (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.

Multi-Party Operations

The Multiple Party Operation recall can only be applied in a standalone environment, and therefore does not interact with this feature.

Network Attendant Service Anti-tromboning

NAS Anti-tromboning is supported by this feature.

Feature packaging

- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145, or
- ISDN Signaling Link (ISL) package 147
- Network Attendant Service (NAS) package 159.

Feature implementation

Task summary list

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

1. Table 30: LD 15 on page 151

   Enable Slow Answer Recall Enhancement for the customer.
2. Table 31: LD 15 on page 152

   Configure Timers for Slow Answer Recall for Transferred External Trunks.

Table 30: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and Options data block

Prompt	Response	Description
...		
- OPT	(SLD) SLA	Slow Answer Recall Enhancement (denied) allowed

Table 31: LD 15

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	TIM	Timers
...		
- RTIM	xxx yyy zzz	Recall timers for Slow Answer, Camp-on and Call Waiting, where: xxx = 0-(30)-378 for Slow Answer yyy = 0-(30)-510 for Camp-on, and zzz = 0-(30)-510 for Call Waiting. These timers indicate in seconds the elapsed time before attendant recall. Slow Answer must be a multiple of six seconds. To change one timer, all three fields must be input.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 19: Software Licenses

Contents

This section contains information on the following topics:

Feature description

Software Licenses provide flexibility and control over system configuration and implementation. Software ordering and pricing is based on the total count of used License parameters.

There are three categories of Licenses:

- Service License
- System License
- Preset License

Key codes control the increments provided for Service- and System-type Licenses when filling customer orders. The License increments are the same globally and are common to all system types. Some regions do not use certain License parameters. For example, North America does not use DECT Visitors Licenses. If this License parameter is introduced to North America, it will be in accordance with the Global Software Structure.

Service License

Service License parameters are chargeable. They are aligned with the two service packages, Enhanced and Premium Service Package.

The Customer/Distributor must first select the service package and then the number of Service License parameters for each type. All Service License parameters must be ordered and filled at the same service package. Combinations or mixtures of License parameters of different service packages are not allowed. For example, do not order Enhanced Package analog License parameters for a customer who is at Premium Service Package. This customer must be serviced with all License parameters at the Premium Package.

Service License parameters use the same Global code for all systems and regional service packages. They are assigned incremental defaults, minimum order quantity amounts, and incremental amounts.

Service License parameters allow a customer to operate at a specified software service package. For example, when License parameters are ordered for Premium Service package, the order management systems automatically provide the customer with the appropriate regional software feature package content.

When allowable limits are exceeded, any additional entry is blocked, and an error message is shown every time a subsequent entry is attempted.

System License

System License parameters are chargeable. They are applicable to the complete system and are not dependent on which service level the customer is using.

Note:

The same order code is used for all system types and regions.

Preset License

Preset License parameters are preset at their maximum in the Avaya factory. These Licenses are non-chargeable. Setting these License parameters to the maximum allows customers to configure License parameters to meet their feature and configuration needs.

Note:

TNs are not used to control user capacity, but are set to the maximum amount for all system types.

Phantom TNs and Virtual TNs

There are two categories of Loops. They are:

- Physical Loops
- Non-physical Loops

For more information about the configuration and capacities for Loops on the different system types, see *Avaya Communication Server 1000M and Meridian 1 Large System Planning and Engineering* (NN43021-220), and*Avaya Communication Server 1000E Planning and Engineering* (NN43041-220).

Physical Loops

Physical Loops are comprised of physical hardware (network cards and the line cards and trunk cards associated with them).

Non-physical Loops

Non-physical Loops exist in software with no physical hardware associated with them. There are two types of non-physical Loops:

- Phantom Loops
- Virtual Loops

Phantom Loops

Use Phantom Loops for the following:

- Phantom TNs: Phantom TNs are programmed as 500/2500 TNs (with no physical hardware). They have Default Call Forwarding DNs programmed to redirect calls to physical telephones. This forwarding can be changed by a user from a physical telephone using the Remote Call Forward feature.

- Virtual Sets: There are two types of Virtual Sets.
    - Proprietary analog and digital TNs (programmed as BCS, M2XXX, and M3XXX series TNs). The Telelink Mobility Switch 1 feature introduced this type of TN for use with the Personal Communications Service (PCS).
    - M3900 (Singe Site) Virtual Office Terminals. These are the virtual M3903 and M3904 TNs programmed for use by Virtual Office Workers with the M3900 (Single Site) Virtual Office feature. The Host TNs (physical telephones) that users log on are not considered to be Virtual Sets and they are not programmed on Phantom Loops.

Virtual Loops

Virtual TNs are programmed on Virtual Loops.

The following types of telephones and trunks are included in the count of Virtual TNs:

- Basic IP user
- Temporary IP users
- IP user
- PCA
- H.323 Access Ports
- ITG ISDN trunks
- SIP Access ports
- DECT users

Capacities

A superloop cannot have both Phantom and Virtual TNs configured.

Phantom Loops and Virtual Loops have different densities and are not interchangeable. The valid loop ranges are described in Table 32: LD 97 - Loop ranges and labeling for Phantom and Virtual loops on page 156.

Table 32: LD 97 - Loop ranges and labeling for Phantom and Virtual loops

Type of system	Phantom loop range	Virtual loop range
PBX 61C CS 1000M SG	N0-N156	V0-V156
PBX 81C CS 1000M MG with Fibre Network Fabric	N0-N252	V0-V252
CS 1000E	N0-N252	V0-V252

Type of system	Phantom loop range	Virtual loop range
PBX 11C	N96-N112	V96-V112
Note: Superloops are numbered in increments of 4. **Note:** Loops 96-112 on PBX 11C correlate to cards 61-99 for phantom/virtual TN mapping.		

The theoretical TN limit is 65,535 units, but the TN limit is restricted by the total TN Licenses limit of 32,767 Licenses.

Large systems

Phantom cards on a Phantom Loop are limited to 16 units each. A Phantom superloop has a maximum of 512 TNs.

A virtual card programmed on a Virtual superloop can have a maximum of 32 TNs. A Virtual superloop has a maximum of 1024 TNs.

License parameters

Table 33: License parameters on page 157 lists the License parameters that are available.

Table 33: License parameters

Service Licenses	System License parameters	Preset License parameters
Traditional Telephones	Personal Call Assistant (PCA)	TNs
DECT Users	ITG ISDN Trunks	ACDN
IP Users	H.323 Access Ports	AML
Basic IP Users	SIP Access Ports	BRAND
Temporary IP Users	AST	LTID
DECT Visitor Users	RAN CON	RAN RTE
ACD Agents	MUS CON	Attendant Consoles
	Survivability	BRI DSL
		MPH DSL
		DATA Ports
		Phantom Ports
		Traditional Trunks

Service Licenses	System License parameters	Preset License parameters
		DCH TMDI D-Channels

Licenses and TN configurations

Table 34: License parameters dependencies on TN configuration on page 158 lists License dependencies related to TN configurations.

Table 34: License parameters dependencies on TN configuration

License mnemonic	How a TN is configured
	Service License parameters
Traditional Telephones	This parameter counts analog (500/2500-type) telephones configured in LD 10, including: • analog ACD agents and AST • Line-side T1/E1 devices, used for voice mail systems, voice response units, and trading turrets (LD 10, TYPE 500) • Faxes and modems (LD 10, TYPE 500, CLS FAXA) • Fax Server ports (LD 10, TYPE 500, FTR FAXS) • Phantom TN configured as predictive dialer with CS 1000E only Phantom ports, wireless and CLASS telephones are not counted (LD 10, TYPE 500, WRLS is NO, CLS CNUD and CNAD). This parameter counts CLASS compatible analog (500/2500-type) telephones (LD 10, TYPE 500, CLS CNAA or CNUA). This parameter counts digital telephones, including digital ACD agents and AST, and Predictive Dialer ports configured on IPMG TN. CallPilot ports, data and phantom ports, and Virtual telephones are not counted. CallPilot Mini ports are counted.
DECT Users	This parameter counts DECT telephones (LD 10, TYPE 500, WRLS YES) supporting concentration. Visiting DECT telephones are not counted.
Temporary IP Users	This parameter counts any IP Phones type with the LD 11 prompts NUID/NHTN are configured.

License mnemonic	How a TN is configured
Basic IP Users	This parameter counts IP Phones type 2001 (LD 11, TYPE 2001P2, 1110 and 2033) Overflow : If insufficient “Temporary IP Users” Licenses are available for any IP Phone with “NUID/NHTN”, then any unused “Basic IP users” or “IP Users” Licenses can also be used for the configuration of these IP Phones with an error message generated, recommending the purchase of additional Basic IP/IP user Licenses.
IP Users	This parameter counts IP Phones type 2002, 2210/2211/2212, 2004, 2007, 1120,1140, 1150 and IP Softphone 2050 (LD 11, TYPE 2004P1, 2004P2, 2210, 2211, 2212, 2007, 1120, 1140, 1150, 2002P1, 2002P2, 2050PC, 2050MC, 6120, and 6140). Overflow: If insufficient “Basic IP users” Licenses are available for the IP Phone [2001/1110/2033], then any unused “IP users” Licenses can also be used for the configuration of the IP Phone [2001/1110/2033] with an error message generated, recommending the purchase of additional Basic IP users Licenses.
DECT Visitor Users	This parameter counts Visiting DECT telephones supporting concentration feature (LD 10, TYPE DCS, VSIT YES).
ACD Agents	This parameter counts analog ACD agents (LD 10, TYPE 500, CLS AGTA, FTR ACD), Wireless ACD agents (LD 10, TYPE DCS or 500, WRLS YES, CLS AGTA, FTR ACD), Digital ACD agents, Meridian Integrated ACD ports, Virtual Office host agents, and Internet ACD agents (LD 11, TYPE 2002P1/P2, 2004 P1/ P2, 2050PC, 1120, 1140, 1150 KEY 0 ACD). CallPilot ports are not counted as ACD Agents. CallPilot Mini ports are counted.
Personal Call Assistants (PCA)	This parameter counts Personal Call Assistant data blocks (LD 11, TYPE PCA, KEY 1 HOT P).
ITG ISDN trunks	This parameter counts ITG-i486 Card trunks, ITG-Pentium Card trunks, and ITG Media Card trunks. Voice gateways are not counted (LD 14, TYPE not VGW, XTRK ITG8, MC8, MC32 and not VGW, IPTN NO).
H.323 Access Ports	This parameter counts H.323 IP Trunks (LD 14, TYPE IPTI; RTMB: route number, unit number). The route is configured as H.323 route (LD 16 PCID: H.323).
SIP Access Ports	This parameter counts SIP IP Trunks (LD 14 TYPE: IPTI; RTMB: route number, unit number). The route is configured as SIP route (LD 16 PCID: SIP).
AST	This parameter counts Associated analog telephones (LD 10, TYPE 500, AST YES), Associated analog ACD agents

License mnemonic	How a TN is configured
	(LD 10, TYPE 500, CLS AGTA, FTR ACD, AACD YES), Associated digital and internet telephones, and Associated trunks. The following trunks cannot be associated: MUS, ADM, R232, R422, MCU, MDM, AWR, PAG, DIC, RAN, RCD) CallPilot ports are not counted.
RAN connections (RAN CON)	This parameter counts Broadcasting RAN trunks (LD 14, TYPE RAN).
Music connections (MUS CON)	This parameter counts Broadcasting music connections. Non-broadcasting music trunks are not counted (LD 14, TYPE MUS).1 Music Broadcasting trunk = 64 Music Connections. USED counter value is the maximum number of simultaneously-used music connections since the last sysload.
Survivability	This parameter counts Survivability License usage. (LD 117: SURV cab YES)
Preset License parameters	
TNs	The total number of TNs refers to Terminal Numbers (TNs) configured in LDs 10, 11, 12, 13, and 14. There is no differentiation between signaling, data, and voice channels.
ACD DNs (ACDN)	ACD DNs counts the number of ACD and CDN data blocks (LD 23, TYPE ACD or CDN).
AML	Application Module Links (LD 17, ADAN AML).
Brand	Brand index License specifies a string of alphanumeric characters displayed on an idle telephone.
LTID	Logical terminals configured on DSLs (LD 27, TYPE DSL).
RAN route	Recorded Announcement Routes (LD 16, TKTP RAN).
Attendant consoles	This parameter counts every attendant console and PC console configured in LD 12. An attendant console can use two or more TNs. However, the TNs occupied by an attendant console are not used for attendant console License counting criteria; each TN occupied by an attendant console is used for System TN License counting criteria. TNs used for power supply are not counted toward attendant consoles.
BRI DSL	This parameter counts every BRI line (LD 27, TYPE DSL, APPL BRIL).
MPH DSLs	This parameter counts every BRI MPH line (LD 27, TYPE DSL, APPL MPH).

License mnemonic	How a TN is configured
Data ports	This parameter counts every Data Port configured in LD 10 (data TNs), LD 11 (data TNs) or LD 14 (MCA, MCU). Data Ports are excluded from counting as Analog or Digital Telephones or Traditional Trunks. A data TN configured in LD 11 is a Data Port. A Meridian Communication Adapter (MCA) fits inside a Meridian Digital Telephone to provide access to data functions. An MCA is configured in LD 11 as an M2006, M2008, M2216 or M2616 or M3900 series with DTAO prompt set to either Meridian Programmable Data Adapter (MPDA) or MCA. A Meridian Communication Unit (MCU) replicates the functionality of the MCA and provides additional features. Both MCA and MCU are counted as Data Ports. A Data Access Card (DAC) is a data interface card that allows the card to work with the RS-232 interface, the RS-422 interface, or both. Configuration of DAC is in LD 11, with R232 or R422 as the TYPE prompt. Both R232 and R422 data terminals are counted as Data Ports. A Data Port is not limited to units 16–31. If a TN has Flexible Voice/Data Allowed (FLXA) CLS, a DATA port is allowed on TN (unit 0-15). ATA terminals (LD 11, TYPE: any of M3xxx, CLS DTA and not MMA) Meridian Communication Adapters (LD 11, TYPE: any of M2xxx, CLS DTA and not MMA) Meridian Communication Units (LD 11, TYPE MCU) R232 DAC units (LD 14, TYPE R232) R422 DAC units (LD 14, TYPE R422).
Phantom ports	Analog phantom telephones configured on Phantom loops (LD 10, TYPE 500). Digital phantom ports configured on Phantom loops (LD 11, TYPE: any of M2xxx and M3xxx).
Traditional Trunks	This parameter counts each Traditional Trunk (analog, digital, ISDN, and ITG 1.0 Trunks) configured in LD 14. Analog trunks that use in-band signaling for establishing calls to COs or other switches are counted as Traditional Trunks. Trunks of this nature include, but are not limited to, the following: • Automatic Identification of Outward Dial (AIOD) • Common Control Switching Arrangement (CCSA) • Automatic Number Identification (ANI) • Autovon (ATVN) • Central Automatic Message Accounting (CAMA) • Central Office (COT) • Common Control Switching arrangement (CSA)

License mnemonic	How a TN is configured
	• Direct Inward Dial (DID) • Foreign Exchange (FEX) • Feature Group D (FGD) • Release Link Main (RLM) • Release Link Remote (RLR) • TIE • Wide Area Telephone Service (WAT) Counting analog trunks does not depend on hardware type, density or country-specificity. DTI channels (1.5 and 2.0 Mb) and JDMI trunks count as Traditional Trunks. Line-Side T1/E1 are counted as Analog Telephones and are not counted as Traditional Trunks. ISDN trunks such as ISL, VNS, 1.5 and 2.0 Mb PRI (including IDA) and BRI count as Traditional Trunks.
D-channels (DCH)	Primary D-channels (LD 17, ADAN DCH) Backup primary D-channels (LD 17, ADAN BDCH) For CS 1000E (Rls 5.0 and later) also applicable to D-channels configured on the TMDI card. This parameter is specific to CS 1000E systems (LD 17, TYPE ADAN, CTYP TMDI).
TMDI D-channels	D-channels configured on the TMDI card. This parameter is specific to PBX 11C system only.

Table 35: TN Set (IP) Type (Model)

Rls 5.0 and later	Rls 4 and Rls 4.5	Applicable IP License			Notes
		IP	Basic	Temp.	
2004P1, 2004P2	I2004	X		X	P1 : Phase 1 & P2 Phase II
2210, 2211, 2212, Avaya 6120 WLAN Handset, Avaya 6140 WLAN Handset	I2004	X		X	WLAN Handset
Avaya 2007 IP Deskphones	I2004	X		X	
Avaya 1140 IP Deskphone	I2004	X		X	
2050PC, 2050MC	I2050	X			I2050 with PCA_OPT flag set (PCA TN) are automatically converted to new TN type PCA

Rls 5.0 and later	Rls 4 and Rls 4.5	Applicable IP License			Notes
		IP	Basic	Temp.	
2002P1, 2002P2	I2002	X		X	
Avaya 1120 IP Deskphone	I2002	X		X	
2001P2	I2001		X	X	
Avaya 1110 IP Deskphone	I2001		X	X	
Avaya 2033 IP Conference Phone	I2001		X	X	Polycom Conference
Avaya 1150 IP Deskphone	IPACDI V		X		“IPACD” mnemonic is changed to “1150”, but no new set type is added

Service Level License default settings and increment values

Table 36: Service Level License default and increment values on page 163 lists Service Level License default settings and increment values.

Table 36: Service Level License default and increment values

License mnemonic	New system default setting by system type	Order increment for new systems and expansions	License ordering guidelines
Traditional telephones	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 0 - CS 1000M-SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/81C 1 - CS 1000M-SG/ MG 1 - CS 1000E	See note 2 below. Provision 8 for CallPilot Mini.
DECT users (previously called Wireless user)	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 0 - CS 1000M-SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/81C 1 - CS 1000M-SG/ MG N/ A- CS 1000E	Previously called Wireless user. Changed in Rls 4.0. See notes 1 and 2 below.
IP users	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 0 - CS 1000M-SG/MG 0 - CS 1000E	N/A - PBX 11C Cabinet/ Chassis N/A - PBX 61C/ 81C 1 - CS 1000M-SG/ MG 1 - CS 1000E	See note 2 below.
Temporary IP users	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 0 - CS 1000M-SG/MG 0 - CS 1000E	N/A - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 1 - CS 1000M-SG/ MG 1 - CS 1000E	License for Rls 5.0 and later. Provides access for IP redundant

License mnemonic	New system default setting by system type	Order increment for new systems and expansions	License ordering guidelines
			set, up to the limit of the License. Initial order is 100 units.
Basic IP users	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 0 - CS 1000M-SG/MG 0 - CS 1000E	N/A - PBX 11C Cabinet/ Chassis N/A - PBX 61C/ 81C 1 - CS 1000M-SG/ MG 1 - CS 1000E	See note 2 below.
DECT Visitor users	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/81C 0 - CS 1000M-SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/81C 1 - CS 1000M-SG/ MG N/A - CS 1000E	This License is only used in the EMEA and Asia Pacific regions. See note 1 below.
ACD Agents	10 - PBX 11C Cabinet/ Chassis 10 - PBX 61C/81C 10 - CS 1000M-SG/MG 10 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/81C 1 - CS 1000M-SG/ MG 1 - CS 1000E	10 ACD Agent users are provisioned for all new system types in all regions. Applicable for any service level ordered.

Note:

For North America and CALA, use DECT User Licenses for upgrades, expansions and transfers of Companion-enabled systems only. The DECT User License is not supported on North American CS 1000E and Media Gateway 1000B systems. North American and CALA Companion DECT Licenses can be moved from an existing pre-Rls 3.0 system to the CVSD structure using OrderPro. DECT User Licenses are used in EMEA and Asia Pacific to support the DECT Wireless product on all products and the Media Gateway 1000B systems (Branch Office).

Note:

CS 1000M SG = CS 1000M Single Group system and CS 1000M MG = CS 1000M Multi Group system

System Level License default settings and increment values

Table 37: System Level License default and increment values on page 165 lists System License default settings and increment values for new systems.

Table 37: System Level License default and increment values

License mnemonic	New system default setting by system type	Order increment for new systems and expansions	License ordering guidelines
Personal Call Assistant (PCA)	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/ 81C 1 - CS 1000M SG/ MG 1 - CS 1000E	These increments apply to standalone Meridian 1, Avaya CS 1000M, Avaya CS 1000E systems.
ITG ISDN Trunks	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M SG/MG 0 - CS 1000E	8 - PBX 11C Cabinet/ Chassis 8 - PBX 61C/ 81C N/A - CS 1000M SG/ MG N/A- CS 1000E	ITG ISDN Trunk can be co-resident in a configuration with Virtual Trunk ports (H.323 Access Ports and SIP Access Ports) for the CS 1000M systems.
H.323 Access Ports	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M SG/MG 0 - CS 1000E	N/A - PBX 11C Cabinet/ Chassis N/A - PBX 61C/ 81C 1 -CS 1000M SG/ MG 1 - CS 1000E	
SIP Access Ports	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M- SG/MG 0 - CS 1000E	N/A - PBX 11C Cabinet/ Chassis N/A - PBX 61C/ 81C 1 -CS 1000M SG/ MG 1 - CS 1000E	
AST	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/ 81C 1 - CS 1000M - SG/MG 1 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/ 81C 1 - CS 1000M SG/ MG 1 - CS 1000E	This License controls Avaya and third-party applications.
RAN CON	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M - SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/ 81C 1 - CS 1000M SG/ MG 1 - CS 1000E	
MUS CON	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M- SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis 1 - PBX 61C/ 81C 1 - CS 1000M SG/ MG 1 - CS 1000E	
Survivability	0 - PBX 11C Cabinet/ Chassis 0 - PBX 61C/ 81C 0 - CS 1000M- SG/MG 0 - CS 1000E	1 - PBX 11C Cabinet/ Chassis N/A - PBX 61C/ 81C N/A - CS 1000M SG/ MG N/A - CS 1000E	

Factory preset License values

Table 38: Factory preset License values on page 166 lists factory preset License values.

Table 38: Factory preset License values

License mnemonic	Value setting by system type
TNs	2500 - PBX 11C Cabinet/Chassis 32760 - PBX 61C/81C 32760 - CS 1000 M - SG/MG 32760 - CS 1000E
AML	16 - PBX 11C Cabinet/Chassis 16 - PBX 61C/81C 16 - CS 1000 M - SG/MG 16 - CS 1000E
LTID	0 - PBX 11C Cabinet/Chassis 32760 - PBX 61C/81C 32760 - CS 1000 M - SG/MG 32760 - CS 1000E
ATTENDANT CONSOLES	2500 - PBX 11C Cabinet/Chassis 32760 - PBX 61C/81C 32760 - CS 1000 M - SG/MG 32760 - CS 1000E
MPH DSL	100 - PBX 11C Cabinet/Chassis 64- PBX 61C/81C 64 - CS 1000 M - SG/MG 64 - CS 1000E
PHANTOM PORTS	2500 - PBX 11C Cabinet/Chassis 32760 - PBX 61C/81C 32760 - CS 1000 M - SG/MG 32760 - CS 1000E
DCH	80 - PBX 11C Cabinet/Chassis 254 - PBX 61C/81C 255 - CS 1000 M - SG/MG 255 - CS 1000E
ACDN	300 - PBX 11C Cabinet/Chassis 24000 - PBX 61C/81C 24000 - CS 1000 M - SG/MG 24000 - CS 1000E
BRAND	2 - PBX 11C Cabinet/Chassis 2 - PBX 61C/81C 2 - CS 1000 M - SG/MG 2 - CS 1000E
RAN RTE	500 - PBX 11C Cabinet/Chassis 512 - PBX 61C/81C 512 - CS 1000 M - SG/MG 512 - CS 1000E
BRI DSL	150 - PBX 11C Cabinet/Chassis 10000 - PBX 61C/81C 10000 - CS 1000 M - SG/MG 10000 - CS 1000E
DATA PORTS	2500 - PBX 11C Cabinet/Chassis 32760 - PBX 61C/81C 32760 - CS 1000 M - SG/MG 32760 - CS 1000E
TRADITIONAL TRUNKS	2500 - PBX 11C Cabinet/Chassis 32760 - PBX 61C/81C 32760 - CS 1000 M - SG/MG 32760 - CS 1000E

License mnemonic	Value setting by system type
TMDI D-CHANNEL	64 - PBX 11C Cabinet/Chassis N/A - PBX 61C/81C N/A - CS 1000 M - SG/MG N/A - CS 1000E

Maximum License limits

Table 39: Maximum License limits on page 167 lists maximum License limits.

Table 39: Maximum License limits

Licenses	PBX 11C	CS 1000E systems	PBX 61C/81C CS 1000M SG/MG
Service Licenses			
Traditional telephones	2 500	32 760	32 760
Temporary IP users	N/A	32 760	32 760
Basic IP users	N/A	32 760	32 760
IP users	N/A	32 760	32 760
DECT users	2 500	32 760	32 760
DECT Visitor users	2 500	10 000	10 000
ACD Agents	1 000	32 760	32 760
System Licenses			
ITG ISDN Trunks	2 500	32 760	32 760
H.323 Access ports	N/A	32 760	32 760
SIP Access Ports	N/A	32 760	32 760
Personal Call Assistant (PCA)	1 248	32 760	32 760
AST	1 000	32 760	32 760
RAN_CON	1 000	32 760	32 760
MUS_CON	1 000	10 000	10 000
Survivability	4	N/A	N/A
Preset Licenses			
TNs	2 500	32 760	32 760
ACDN	300	24 000	24 000
AML	N/A	16	16

Licenses	PBX 11C	CS 1000E systems	PBX 61C/81C CS 1000M SG/MG
LTID	2 500	32 760	32 760
RAN_RTE	500	512	512
BRAND	2	2	2
Attendant Consoles	2 500	32 760	32 760
BRI_DSL	150	32 760	32 760
MPH_DSL	N/A	32 760	32 760
Data Ports	2 500	32 760	32 760
Phantom Ports	2 500	32 760	32 760
Traditional Trunks	2 500	32 760	32 760
DCH	N/A	255	255
TMDI_D-Channel	64	255	N/A
Note: The values presented in this table are individual License limits, not engineering limits or rules.			

System monitoring

To assist in monitoring system growth, each time an overlay is used, a header appears in the affected overlay, reflecting the system status. The header indicates the total, available, and used quantities of the License parameters corresponding to the data blocks that are configured in the overlay. The counts are updated each time system activity adds or deletes one of the tracked items. When the limits are exceeded, an error message appears.

ACD parameters are preset for each system. The numbers in the header are not necessarily real limits and are subject to system configuration. Contact your Avaya representative for information regarding your system limits.

A header, reflecting License parameters, is present in the following overlays:

- LD 10: DECT (500/DCS) telephones, DECT visitors, ACD agents, AST, TNs, data ports and phantom ports.
- LD 11: Meridian 1 proprietary telephones, IP Phones, ACD agents, PCAs, AST, TNs and data ports.
- LD 12: Attendant Consoles and the number of TNs.
- LD 13: Digitone receivers and tone detectors
- LD 14: AST, ITG ISDN trunks, H.323 access ports, SIP access ports, RAN and MUS connections, TNs, data ports and traditional trunks.

- LD 16: RAN routes
- LD 17: D-channels (DCH and TMDI DCH) and Application Module Links (AMLs)
- LD 23: ACD-DNs
- LD 27: TNs, Digital Subscriber Loops (DSLs) and Logical Terminal Identifiers (LTIDs).
- LD 117: Survivability

Printing system License limits

When REQ is set to SLT in LD 22, system License limits are printed. You can update the value of License limits either through sysload or the Instant Software License feature. You can print the new License limits through LD 22 after the update is complete.

The LD 22 implementation for printing system limits is as follows:

Table 40: LD 22 - Print system limits

Prompt	Response	Description
REQ	SLT	Print System Limits: Incremental Software Management.

In the License limits printout, three parameters are printed for each License:

- The first parameter is the License limit.
- The USED parameter is the number of configured units.
- The LEFT parameter is the difference between the License limit and the USED value (LEFT = License limit - USED).

```
TRADITIONAL TELEPHONES        800   LEFT 784   USED 16
DECT USERS                    0     LEFT 0     USED 0
IP USERS                      500   LEFT 0     USED 500
BASIC IP USERS                60    LEFT 0     USED 60      Overflow 20
TEMPORARY IP USERS            100   LEFT 0     USED 100     Overflow 20
DECT VISITOR USER             0     LEFT 0     USED 0
ACD AGENTS                    300   LEFT 189   USED 111


PCA                           20    LEFT 9     USED 11
ITG ISDN TRUNKS               0     LEFT 0     USED 0
H.323 ACCESS PORTS            30    LEFT 0     USED 30
AST                           32    LEFT 0     USED 32
SIP CONVERGED DESKTOPS        0     LEFT 0     USED 0
SIP CTI TR87                  0     LEFT 0     USED 0
SIP ACCESS PORTS              250   LEFT 0     USED 250
RAN CON                       120   LEFT 0     USED 120
MUS CON                       120   LEFT 0     USED 120


TNS                           32767 LEFT 31444 USED 1323
ACDN                          24000 LEFT 23998 USED 2
AML                           16    LEFT 15    USED 1
IDLE_SET_DISPLAY NODE D CP-PM CS 491L/11
LTID                          32760 LEFT 32760 USED 0
RAN RTE                       512   LEFT 510   USED 2
ATTENDANT CONSOLES            32767 LEFT 32766 USED 1
BRI DSL                       10000 LEFT 10000 USED 0
MPH DSL                       100   LEFT 100   USED 0
DATA PORTS                    32767 LEFT 32767 USED 0
PHANTOM PORTS                 32767 LEFT 32767 USED 0
TRADITIONAL TRUNKS            32767 LEFT 32691 USED 76
DCH                           255   LEFT 252   USED 3
```

Figure 9: Example of a LD 22 printout (REQ=SLT).

Note:

For Music Broadcast connections (MUS CON), the USED parameter is the maximum License usage since the last sysload.

System administration

When the predefined License limits are reached, an error message indicates that further database additions are blocked.

A new keycode must be ordered to increase system limits. In order to minimize delays in system administration, it is critical that the configuration limits be monitored and that new disks are ordered before the current parameters are exceeded.

Software Upgrade

When performing a system upgrade, if any of the new License limits exceed present limits, then do not attempt to sysload. Excess information will be lost. Obtain new disks with expanded limits.

Caution:

System information will be lost. Upon software upgrade, if SYS message 4327, 4328, 4329, or 4330 appears at SYSLOAD, then SYSLOAD previous system disks. Order License disks with sufficient system parameters configured.

DO NOT DATADUMP; system information will be lost. Call your technical support department for assistance.

Keycodes

A keycode is a machine-generated, digitally-signed list of customer capabilities and authorized software release. A security keycode scheme protects License parameters.

For a customer to expand License limits, they must order and install a new keycode. This installation is performed using the Keycode Management feature. Use LD 143 to run all Keycode Management commands.

There are conditions under which a customer must sysload. For more information, see the Instant License chapter in *Avaya Features and Services Fundamentals — Book 4 of 6 (I to M), NN43001-106*.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

Software Licenses requires the following packages:

- ACD-DNs and ACD Agent
  - Basic ACD (BACD) package 40
- Application Module Link (AML)
  - Digit Display (DDSP) package 19
  - ACD Package B (ACD-B) package 41
  - ACD Package A (ACD-A) package 45
  - Command Status Link package 77
  - ISDN Application Module Link for Third Party Vendors (IAP3P) package 153
- AST
  - Command Status Link (CSL) package 77
  - Application Module Link (AML) package 209
- Attendant Consoles
  - Attendant Consoles is included in base system software.
- CLASS Telephones
  - Calling Number Delivery (CNUMB) package 332 or
  - Calling Name Delivery (CNAME) package 333
- Data Ports
  - Package requirements for data ports vary depending on the type of data port configured. See *Avaya Software Input/Output Reference — Administration* (NN43001-611) and *Avaya Software Input/Output Reference — Maintenance , NN43001-711* for information on specific data port package requirements.
- IP Phones
  - M2000 Digital Set (DSET) package 88
  - Aries Digital Set (ARIE) package 170
- ITG ISDN Trunks
  - Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58
  - Integrated Services Digital Network (ISDN) package 145

    - ISDN Signaling Link (ISL) package 147
    - Multi-purpose Serial Data Link (MSDL) package 222 (for Large Systems only)
- Phantom Ports
    - Phantom TN (PHTN) package 254
- Traditional Trunks
    - Package requirements for Traditional Trunks vary depending on the type of trunk configured. For more information about specific trunk package requirements, see *Avaya Software Input/Output Reference — Administration* (NN43001-611) and *Avaya Software Input/Output Reference — Maintenance , NN43001-711*.
- DECT
    - Meridian 1 Companion Option (MCMO) package 240
- Personal Call Assistant (PCA)
    - Personal Call Assistant (PCA) package 398
- H.323 Access Ports
    - IP Peer Networking package 399
- SIP Access ports
    - SIP Gateway and Converged Desktop package 406

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 20: Source Included when Attendant Dials

Contents

This section contains information on the following topics:

Feature description

This feature provides a new option in LD 15, which allows the customer to define whether or not the source is to be included in a call while the attendant is dialing the destination (SIAA = allow, SIAD = deny). If the destination answers while the attendant is still included in the call, intrusion tone is provided to all parties to indicate that a conference has been established. The intrusion tone is defined in LD 56, and is a prerequisite for the Source Included when Attendant Dials feature.

If SIAA has been defined, the source will be included in all situations, regardless of the state of the destination, except when the attendant is performing Break-In to a busy station.

The following table outlines the operation, if SIAA has been defined, according to the state of the destination party:

	Source	
Destination	**Included**	**Excluded**
Idle extension	x	

	Source	
Destination	**Included**	**Excluded**
First Degree Busy	x	
Second Degree Busy	x	
Camp-on	x	
Intercept forwarded	x	
Line lock-out	x	
Vacant	x	
Busy, Attendant Break-in		x
Recorded Announcement	x	
Music	x	

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN feature will follow the current Source Included when Attendant Dialing handling occurs.

Attendant Break-In

The operation of the Break-In feature is not affected, except that the source receives busy tone before the attendant presses the Break-In (BKI) key.

Attendant Supervisory Console

While the attendant dials the destination, the source receives intrusion tone.

Automatic Call Distribution

The source is included in a conference involving the attendant, the source, and Automatic Call Distribution (ACD). When the call is answered by the ACD agent, intrusion tone is provided to all parties in the conference.

Camp-On, Semi-automatic Camp-On

The source remains included while the attendant dials the destination.

Intercept treatment

If the attendant dials a destination which is intercepted, the source remains included in the call.

Recorded Announcement, Music

The source is included in a conference involving the attendant, the source, and Recorded Announcement or music treatment. Intrusion tone is not provided in this case.

Secrecy Enhancement

Source Included when Attendant Dials takes precedence over Secrecy and Enhanced Secrecy.

Feature packaging

This feature requires:

- International Supplementary Features (SUPP) package 131
- Flexible Tone and Cadences (FTC) package 125
- Trunk Barring (TBAR) package 132

Feature implementation

Table 41: LD 15 - Configure Source Included when Attendant Dials for a customer.

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	FTR	Features and Options
...		
- OPT	(SIAD) SIAA	(Deny) or allow Source Included when Attendant Dials.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 21: Special Dial Tones after Dialed Numbers

Contents

This section contains information on the following topics:

Feature description

This feature allows special dial tones to be provided after certain telephone numbers are dialed. Both the telephone numbers and associated dial tones are customer-defined in LD 56. The system can handle a list of up to 20 telephone numbers with a maximum length of five digits. A tone can be associated with each number. Several different tones can be provided during a dialing sequence by defining a tone with any combination of digits in the dialed number.

For example, for the number 12345, a tone can be provided after the digit 1 is dialed, after the digits 123 are dialed, and after the whole (12345) number is dialed. This is done by defining a tone with the digit 1, a tone with the digits 123, and a tone with the digits 12345.

When a number is dialed, the system performs digit analysis. As soon as the dialing sequence is recognized as part of the customer-defined list, the system provides the associated tone, if one has been defined. The tone is generated after all other treatment of digits is performed. As soon as another digit is dialed, the tone is removed. This digit analysis is done until the dialing sequence is completed.

Tones are provided to the following originating terminals:

- all types of sets (including data terminals) and attendants, and
- TIE trunks, except those with MFC/MFE signaling.

Operating parameters

The system performs digit analysis before any other treatment of digits, except digit insertion for incoming trunk calls.

In a network environment, digit recognition is reported to the distant node, which must be equipped to handle the processing.

Feature interactions

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

The Special Dial Tones after Dialed Numbers feature is supported in a DPNSS1 UDP network.

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

Special Dial Tones can be used to provide dial tone after the system user has dialed the digit 9 (Local Exchange access code).

EuroISDN Master Mode

This feature is not supported for incoming calls on the ETSI network side, but it is supported for outgoing calls.

Special dial tone after access codes

Special dial tone after access codes takes precedence over the special dial tones after dialed number treatment. To define special dial tones after access codes, NO has to be entered in response to prompt DLTN in LD 86 (to inhibit dial tone to access codes). The access code digits and associated tones would then have to be defined in response to the DTAD prompt in LD 56.

Feature packaging

Flexible Numbering Plan (PNP) package 160; and to define SRC1-SRC8 special tones, the Flexible Tones and Cadences (FTC) package 125.

Feature implementation

Task summary list

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

1. Table 42: LD 86 on page 181

   Enable Special Dial Tones after Dialed Numbers.
2. Table 43: LD 15 on page 182

   Configure Special Dial Tones after Dialed Numbers.

Table 42: LD 86

Prompt	Response	Comment
REQ	CHG	Change
CUST	xx	Customer number, as defined in LD 15
FEAT	ESN	ESN (Electronic switched network)
DLTN	(YES) NO	NARS/BARS Dial Tone after dialing AC1 or AC2 access codes.

Table 43: LD 15

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	DTAD	Special Dial Tone after Dialed Number data block.
DDGT ...	x...x	Dialed digits (1-5 digits).
- TONE	a...a	Tone to be provided after the dialed digits Where a...a: (DIAL) = Dial Tone SPDT = Special Dial Tone SRC!-SRC8 = Source tones 1-8 (Valid if FTC package 125 is equipped)

Feature operation

No specific operating procedures are required to use this feature.

Chapter 22: Special Signaling Protocols

Contents

This section contains information on the following topics:

Feature description

This feature allows the existing Swedish analog (500/2500-type) telephones to be connected through analog TIE trunks to the system. These TIE trunks use Swedish signaling protocols. The TIE trunks can be divided into the following types:

- automatic
- semi-automatic
- tone, or
- Automatic Telephony (ATL) (when the Swedish ATL trunk support feature is equipped).

Operating parameters

The Swedish TIE trunk types do not apply to digital TIE trunks.

The Swedish TIE trunk types cannot be mixed on a route.

The Swedish TIE trunks require trunk cards of type TPC71 or TPC237. The trunk cards must be placed on specific Televerket (TVT) loops.

A semi-automatic or tone TIE trunk should not be connected to another system trunk. An incoming Public Exchange/Central Office trunk can be connected to an outgoing automatic TIE trunk.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 44: LD 16 - Configure Trunk route for Special Signaling Protocols.

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route data block
CUST	xx	Customer number, as defined in LD 15
...		
TKTP	TIE SEMI TIE AUTO TIE TONE	Semi-automatic TIE trunk data block. Automatic TIE trunk data block. Tone TIE trunk data block.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 23: Special Trunk Support

Contents

This section contains information on the following topics:

Feature description

This feature allows the interface of the system with the Swedish Automatic Telephony (ATL) military radio-link network.

Operating parameters

ATL trunks must never be used for tandem switching or for networks using Electronic Switched Network (ESN) proprietary signaling.

Echo suppression and loss adjustment cannot be effected through software change.

Modified TPC237 cards must be used for ATL trunks, and must be configured on loops specifically defined for Televerket (TVT) use. An SSO adapter is used between the ATL network trunk and the TPC237 card.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Task summary list

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

1. Table 45: LD 14 on page 186

   Configure Trunks for Special Trunk Support.
2. Table 46: LD 16 on page 187

   Enable Trunk Routes for Special Trunk Support.

Table 45: LD 14

Prompt	Response	Comment
REQ	CHG	Change
...		
TYPE	TIE	TIE Trunk data block.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		
CUST	xx	Customer number, as defined in LD 15
...		

Prompt	Response	Comment
NCOS	(0)	Network Class of Service.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
...		
MNDN	9	Manual Directory Number.
TGAR	(0)	Trunk Group Access Restriction.
SIGL	EAM	Trunk signaling. E&M two-wire.
...		
STRI	WNK	Wink or Fast Flash.
STRO	WNK	Wink or Fast Flash
SUPN	YES	Answer and disconnect supervision required.

Table 46: LD 16

Prompt	Response	Comment
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	TIE ATL	The ATL data block for Sweden.
...		
ICOG	IAO	Incoming and outgoing trunk.
...		
SRCH	RRB	Round Robin Hunting for outgoing trunk (start with the next lower trunk than the one seized).
...		
ACOD	xxxx	Access Code for the trunk route. The ACOD must not conflict with the numbering plan.
...		
CNTL	YES	Change controls or timers.

Prompt	Response	Comment
- TIMR	ODT 8064	End of dial tone for Digitone trunks in milliseconds.
- TIMR	EOD 8064	End of Dial, non-Digitone trunks in milliseconds.
- TIMR	DSI 20096	Disconnect Supervision in milliseconds.
- TIMR	ICF 896	Incoming Flash in milliseconds.
- TIMR	OGF 896	Outgoing Flash in milliseconds.
- TIMR	GTI 1152	Incoming Guard in milliseconds.
- TIMR	GTO 1152	Outgoing Guard in milliseconds.
- TIMR	OBA 120	Outgoing B-Answer. Time in seconds to wait for B-Answer on outgoing ATL trunks for Sweden.
- SST	4	Seizure Supervision Timer, in seconds.
NEDC	ETH	Either end control.
FEDC	ETH	Far End Disconnect Control. Either end.
...		
PANS	YES	Pseudo Answer can be sent on some types of trunks as soon as end of dialing is detected. SUPN in LD 14 should be YES, or PANS = YES has no meaning.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 24: Speed Call Delimiter

Contents

This section contains information on the following topics:

Feature description

The Speed Call Delimiter feature meets the Chinese Ministry of Posts and Telecommunications requirements for the operation of Speed Call and System Speed Call. This feature operates similar to the Speed Call and System Speed Call with the exception of delimiters and confirmation tones.

The Speed Call Delimiter feature requires a Speed Call controller to enter an asterisk (*) between abbreviated numbers and telephone numbers when configuring speed call lists. An additional octothorpe (#) delimiter is required for Analog (2500-type) sets to indicate the end of dialing. If an octothorpe (#) is not entered, then the telephone number is not stored and the entry is not valid.

The octothorpe (#) delimiter has the flexibility of being programmed as mandatory or optional. The delimiter can be modified to something other than an octothorpe (#).

Operating parameters

An asterisk (*) delimiter is used when programming speed call lists only. An asterisk (*) can also be used as a three second delay.

No changes occur when a user wants to display a number stored against a list entry number. To display a stored entry the user presses the Display key and the Speed Call key and dials the list number. The list number cannot be abbreviated.

This feature does not apply to Analog 500-type telephones.

The use of confirmation tone or announcement implies the use of an (#) as end of dial speed call delimiter. This means that an (#) cannot be stored as part of the digit string.

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network access code and dialed number are part of the Speed Call or Autodial Key. If the (#) is not entered, then the user receives a fast busy tone. Therefore if MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe at the FFCS prompt in LD 15.

Feature interactions

Autodial, Speed Call

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Group Call List

Speed Call Delimiter does not interact with Group Call List.

Outpulsing Asterisk and Octothorpe

If the Outpulsing Asterisk (*) and Octothorpe (#) (OPAO) package 104 is equipped and the configuration tone is programmed, then the value stored in the STRG prompt (LD 15) is entered rather than an octothorpe (#) to indicate the end of dial string. Following this, the numbers are stored.

Feature packaging

China Speed Call Delimiter requires Speed Call (OBTF) package 1 and System Speed Call (SSC) package 34.

Flexible Feature Codes (FFC) package 139 is required for Analog 2500-type telephones, if a telephone accesses speed call list or system speed call list or attendant console. This package is optional for proprietary sets or attendant consoles because these sets can access Speed Call List/System Speed Call List by using a key.

Feature implementation

To enable Speed Call and System Speed Call, the maximum number of speed call lists must be determined in LD 17. The speed call list memory size must also be configured in LD 18. For more information on these overlays and the assignment of these features to proprietary, analog (500/2500-type) telephones and attendant consoles refer to the sections entitled Speed Call and System Speed Call in this publication.

Table 47: LD 15 - Enable Speed Call Delimiter in Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Features and options
...		
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		
- LEND	YES	List Entry Number Delimiter. If LEND=YES, then an asterisk (*) delimiter between the list entry number and

Prompt	Response	Description
		telephone number must be entered. If LEND=NO, then existing Speed Call operation continues.
- MSCD	YES	Mandatory Speed Call Delimiter. Default = Octothorpe (#). An octothorpe (#) is required after entering telephone number to indicate the end of dial. If MSCD=NO, then the end of dial Speed Call Delimiter octothorpe (#) is optional.
The China market requires an octothorpe (#) delimiter at the end of dialing. Other markets have the option of selecting a mandatory or optional delimiter by entering YES or NO at the MSCD prompt. The end of dial delimiter can be an octothorpe (default value) or it can be changed to another delimiter by modifying values at the Flexible Feature Code end-of-dialing indicator (FFCS). String to indicate end-of-dialing (STRG) and string length of end-of-dial indictor (STRL) prompts in LD 15.		

Feature operation

Speed Call Delimiter Operation

Analog 2500-type telephone

- To Program Speed Call List — Go off-hook, dial and receive dial tone. Dial System Speed Call Controller (SCC) FFC code, the list entry number and telephone number (for example, *51*1*5556667777#). Get response. If accepted, then confirmation tone or announcement is configured and the end of dial speed call delimiter is entered. Response is a tone or speech signal. Otherwise, silence is given. Go on-hook.
- To use — Go off-hook and receive dial tone. Dial Speed Call User (SCU) code and the list entry number.
- To delete List Entry Number in Speed Call List — Go off-hook and receive dial tone. Dial Speed Call Erase (SCE) FFC followed by list entry number (0 - 999) and (#) delimiter. Delete the specific list entry number.

Proprietary telephones and attendant console

- To program Speed Call List — Press Speed Call Controller Key and the indicator flashes. Dial list entry number (0 - 9999) followed by an asterisk (for example, 1*5556667777). Press Speed Call Controller Key again. If entry is accepted, the indicator goes off. If the entry is not accepted, then the indicator remains flashing. An asterisk is only used to indicate the end of dial of list entry number and is not stored as a digit string.

- To use on proprietary telephones — Lift the handset or press DN Key. Press System Speed Call Controller (SCC) or Speed Call User (SCU) Key. Dial the list entry number.
- Use attendant console — Press an idle loop key and then press Speed Call Controller (SCC) Key. Dial the list entry number.

System Speed Call Delimiter Operation

Proprietary telephones

- To Program System Speed Call List — Press assigned System Speed Call Controller Key and indicator flashes. Dial list entry number (0 - 999) followed by an asterisk (*) and then the telephone number (for example, 1*0115556667777). Then press SSC/SSU Key again. If accepted, the indictor goes off. If not accepted, the indicator remains flashing.
- To use — Lift handset or press DN key. Press SSC/SSU Key. Dial the list entry number, or lift handset or press DN key. Dial SSU FFC code. Dial list entry number.

Attendant console

- To program System Speed Call List — Press SSC Key and indicator flashes. Dial list entry number (0 - 999), followed by an asterisk (*) and the telephone number. Press SSC Key again. If entry is accepted, indicator goes off. If the entry is not accepted, the indicator continues to flash.
- To use — Press an idle loop Key. Dial SSU FFC code. Dial list entry number.

Chapter 25: Speed Call Directory Number Access

Contents

This section contains information on the following topics:

Feature description

The Speed Call Directory Number (DN) Access feature allows a Pilot DN to be used as an access code to either a Speed Call List (SCL) or a System Speed Call List (SSC).

Speed Call DN Access provides an alternative way to access either a Speed Call List or a System Speed Call List. Instead of dialing the Special Prefix (SPRE), a SCL or SSC access code, and a list entry number, or instead of depressing an idle DN key, a SCL or SSC key, and then dialing a list entry number, a user can alternatively dial a speed call access Pilot DN followed by the list entry number.

Since each speed call access Pilot DN is associated with a SCL or SSC list, users can access as many SCL or SSC lists as they need by dialing the appropriate Pilot DN.

A Pilot DN can be accessed from anywhere in a network, so that any network user can access all speed call lists defined for a network, from anywhere in the network. This allows a centralized Speed Call List to be set up for the entire network.

Operating parameters

The requirements for Speed Call and System Speed Call also apply to this feature.

Feature interactions

Direct Inward Dialing (DID) and TIE trunk access

An additional one to three digits will be accepted from these trunks to complete a Speed Call, provided these additional digits are allowed to be sent by the external system.

Speed Call, System Speed Call

Speed Call DN Access is an enhancement of the SCL and SSC features. Refer to SCL and SSC feature descriptions for interactions with other features.

Feature packaging

Speed Call Directory Number Access requires Group Hunt/DN Access to SCL (PLDN) package 120.

Dependencies:

- International Supplementary Features (SUPP) package 131
- Flexible Feature Codes (FFC) package 139
- System Speed Call (SSC) package 34
- Optional Features (OPTF) number 1

Feature implementation

Use Table 48: LD 57 on page 197 to define, change, print, or remove data associated with FFC. A new PLDN prompt is introduced for Pilot DNs. The new LSNO prompt is used to associate the Pilot DN with a SCL or SSC list. The USE prompt is displayed only if the Pilot DN entered in response to the PLDN prompt has not already been defined.

Table 48: LD 57

Prompt	Response	Description
REQ	CHG NEW	Modify or create data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	<CR>	Flexible Feature Confirmation Tone.
CODE	PLDN	Code to be modified or created: Pilot DN.
PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be modified or created; enter carriage return to proceed to next prompt.
USE	SCLC SCLU	USE: enter USE for Pilot DN. Speed Call List Controller. Speed Call List User.
LSNO	xxxx	List Number: enter Speed Call or System Speed Call list number. Speed Call list must exist in LD 18.

Prompt	Response	Description
REQ	OUT PRT	Remove or print a code or data block.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
CODE	PLDN ALL	Code requested: Pilot DN. All FFC.
PLDN	xxxx <CR>	Pilot DN: enter Pilot DN to be removed enter carriage return to proceed next prompt

Feature operation

To access either a Speed Call List or a System Speed Call List using this feature, dial a speed call access Pilot DN followed by the list entry number.

Pilot DN

Pilot DNs are defined as PLDN Flexible Feature Codes (FFC) via service change LD 57.

Pilot DNs can be used in two ways:

1. If the USE prompt is set to GPHT, the Pilot DN is defined to activate Group Hunting.
2. If the USE prompt is set to SCLC (Speed Call List Controller) or SCLU (Speed Call List User), the Pilot DN is defined to access the Speed Call or System Speed Call lists that are associated with the Pilot DN.

When the response to the USE prompt is SCLC (controller), a station can modify an SCL or SSC list by dialing the speed call access Pilot DN associated with that list, followed by a one- to three-digit list entry number, the number to be entered in the list, and then going on-hook.

Overflow tone is returned if the information entered is not valid. Confirmation tone is returned if the Flexible Feature Confirmation Tone (FFCT) option is set and trailing '#' is dialed, as in existing Flexible Feature Codes (FFCs) operations.

When the response to the USE prompt is SCLU (user), to use any entry in a SCL or SSC list, a station user dials the speed call access Pilot DN associated with the list, followed by the one- to three-digit list entry number.

Chapter 26: Speed Call on Private Lines

Contents

This section contains information on the following topics:

Feature description

This feature allows Meridian 1 proprietary telephone users equipped with a Private Line (PVR or PVN) key and a Speed Call (SCL) key to first access a Private Line trunk (by pressing the PVR or PVN key) and then make a speed call (by pressing the SCL key).

Operating parameters

When a Private Line call is made, recognizable Route Access Codes are absorbed from the start of every entry in the Speed Call List (for example, if 7654 is stored as a Speed Call List entry, and 76 is a valid Route Access Code, 76 is absorbed and 54 is outpulsed).

Feature interactions

Automatic Redial, Private Line, Speed Call features

The Automatic Redial (ARDL) feature is activated on a number dialed using the Private Line (PVR/PVN) key and then making a speed call by pressing the Speed Call (SCL) key.

Basic/Network Alternate Route Selection (BARS/NARS)

The BARS and NARS access codes (AC1 and AC2) are not absorbed. If a user has a Speed Call list entry that includes either AC1 or AC2, this entry will not terminate correctly when used on a Private Line. The BARS or NARS access code (AC1 or AC2) will be outpulsed, causing the Public Network to either terminate the call at an unwanted location or reject the call.

Feature packaging

Speed Call on Private Lines is part of base system software and requires Optional Features (OPTF) package 1.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

	ACTION	RESPONSE
1	User presses Private Line Ringing (PVR) or Private Line Nonringing (PVN) key.	Trunk is accessed and dial tone is returned.

	ACTION	RESPONSE
2	User presses Speed Call key and enters list entry number.	The number stored against this entry is outpulsed.

Chapter 27: Speed Call

Contents

This section contains information on the following topics:

Feature description

Speed Call allows you to place calls by dialing a one-, two-, or three-digit code. You can use Speed Call for both internal and external calls. To use Speed Call, Meridian 1 proprietary telephones, and attendant consoles can have a Speed Call key/lamp pair.

Analog (500/2500-type) telephones can activate Speed Call by using Special Prefix (SPRE) or Flexible Feature Codes (FFC).

Analog (500/2500-type) telephones, Meridian 1 proprietary telephones, and attendant consoles can be designated as a Speed Call Controller (SCC) or a Speed Call User (SCU). SCCs can program the numbers to be stored (Speed Call codes) and can use the Speed Call list. SPUs cannot program Speed Call codes; they can only use the Speed Call lists.

Each stored number is assigned a Speed Call code from the Speed Call list. Each list can contain up to 1000 telephone numbers (entries). The maximum number of digits of the telephone number that can be stored in each entry is specified by the customer. Speed Call entries can be 4, 8, 12, 16, 20, 24, 28, or 31 digits long.

Operating parameters

You can define up to 8191 (0-8190) Speed Call lists per system, as long as sufficient memory is available. The maximum includes all combined Speed Call, System Speed Call (SSC), and Hot Line lists.

You can have as many Speed Call lists as you have available key/lamp pairs on any Meridian 1 proprietary telephone, or attendant console. Any number of users can be assigned to a list. Analog (500/2500-type) telephones can access only one Speed Call list. More than one Speed Call Controller can be assigned to each list, but this is not recommended.

A maximum of 31 digits for the telephone number is allowed per Speed Call list entry. An asterisk (*), which indicates a pause, and an octothorpe (#), which indicates end-of-dialing, can be programmed as part of the entry.

The asterisk (*) used to introduce a pause while outpulsing digits is supported on analog and DTI trunks, but not supported on ISDN trunks. On ISDN trunks, if the OPAO feature is enabled, the asterisk (*) is outpulsed as a called party digit.

Speed Call list entries can be defined in LD 18 or by Speed Call Controllers. Speed Call Controllers must know the digit length (one, two, or three) required for the Speed Call codes in each list.

Feature interactions

AC15 Recall: Transfer from Meridian 1

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

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

Autodial Tandem Transfer

The Speed Call key cannot be used after a Centrex Switchhook Flash or during an established call to send digits out to the far site. The Speed Call key can be used only during the dialing stage.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using Speed Call (SCL) and System Speed Call (SSU/SSC) keys.

Call Forward/Hunt Override Via Flexible Feature Code

The Call Forward/Hunt Override FFC cannot be stored in a speed call list

Call Park

Speed Call can be programmed to parked calls or access parked calls.

Call Party Name Display

No name information displays during the programming of Speed Call numbers.

Calling Party Privacy

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Charge Account and Calling Party Number

Charge account numbers, including the Charge Account access Special Prefix (SPRE) code, can be stored as Speed Call or Autodial numbers. All current limitations of these features apply, such as a maximum of 23 digits per entry, including the access code. An Autodial number or dialed digits can follow, but not precede, a Speed Call number. The digits generated by an Autodial key during feature operation are accepted as Charge Account digits.

Charge Account, Forced

Forced Charge Account numbers (including the Special Prefix [SPRE] code and the Charge Account access code) can be entered in Speed Call lists or stored as Autodial numbers. The digits can also be stored, provided that the account number, regardless of its length, is followed directly by an octothorpe (#).

Enhanced Flexible Feature Codes - Outgoing Call Barring

Digits dialed using Speed Call are checked against the active Outgoing Call Barring (OCB) level. This includes calls made using the Dial Access to Speed Call feature (that is, using Pilot DNs).

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Speed Call.

Direct Private Network Access

If a Speed Call entry is programmed with a valid Authcode for Authcode Last followed by an octothorpe (#), the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be reprompted for the Authcode.

Last Number Redial

A number dialed using Speed Call will become the Last Number Redial number on all telephones, except the M2317.

Pretranslation

A Speed Call List number should be programmed to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

If Pretranslation is enabled for a customer, then when a Speed Call List is assigned to a Pretranslation group within the customer, it cannot be accessed by a Meridian 1 proprietary telephone from within that customer group.

Scheduled Access Restrictions

The System Speed Call features ignore the Class of Service and TGAR access restrictions in a Scheduled Access Restriction schedule, using the Class of Service and NCOS defined in the speed call list.

Speed Call Delimiter

An octothorpe (#) is required as a delimiter following an authorization code if an Electronic Switched Network (ESN) and dialed number are stored as part of the speed call or autodial key. If an octothorpe (#) is not entered then the user receives a fast busy tone. If the MSCD = YES, then the end of dial delimiter must be programmed to something other than an octothorpe (#) in LD 15.

Speed Call Directory Number Access

Speed Call DN Access is an enhancement of the Speed Call List (SCL) and System Speed Call (SSC) List features. Refer to Speed Call, System on page 213 for interactions with other features.

Station Specific Authorization Code

Station Specific Authorization Code (SSAU) feature treats stored autodial numbers as if they were entered at the telephone.

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

Speed Call on an E3W trunk will fail for toll calls. E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

User Selectable Call Redirection

Speed Call is not supported by User Selectable Call Redirection.

Feature packaging

Speed Call is part of Optional Features (OPTF) package 1, and has no feature package dependencies.

Feature implementation

Task summary list

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

1. Table 49: LD 17 on page 209

   Set maximum number of Speed Call lists.
2. Table 50: LD 18 on page 209

   Determine if there are enough memory and disk records for new Speed Call Lists.
3. Table 51: LD 18 on page 209

   Add a new Speed Call list.
4. Table 52: LD 10 on page 210

   Assign a Speed Call to an Analog (500/2500-type) telephone.
5. Table 53: LD 11 on page 210

Assign a Speed Call list to proprietary telephone.

6. Table 54: LD 12 on page 211

Assign a Speed Call list to an attendant console.

Table 49: LD 17

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	System Parameters Datablock
...		
- MSCL	0 -8190	Maximum number of Speed Call lists.

Table 50: LD 18

Prompt	Response	Description
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8191	Number of lists to be added.
DNSZ	4-(16)-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of entries in Speed Call list.
Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.		

Table 51: LD 18

Prompt	Response	Description
REQ	NEW CHG OUT	Add, change, or remove a Speed Call list.
TYPE	SCL	Speed Call data block.
LNSO	0-8190	Speed Call list number.
DNSZ	4-(16)-31	Maximum number of digits in a list entry (that is, 4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list may be updated.
STOR	xxx yy...yy	xxx = list entry number (0-9, 00-99, or 000-999). yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	(YES) NO	Data is correct and list can be updated.

Prompt	Response	Description
The prompt WRT follows prompts SIZE and STOR, asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating the data was not stored and must be reentered. A response of **** aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved. The following information is output with the WRT prompt, following SIZE: ADDS: MEM: xxxxx DISK: yy.y where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new Speed Call list. Check the MEM AVAIL and DISK REC AVAIL values output before the REQ prompt.		

Table 52: 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.
FTR	SCU yyyy	Speed Call User, list number (0-8190)
	SCC yyyy	Speed Call Controller, list number (0-8190)

Table 53: 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.
KEY	xx SCU yyyy	System Speed Call User key
	xx SCC yyyy	Speed Call Controller key, where: xx = key number, and yyyy = Speed Call list number (0-8190) M2317 must use key 0-10 or key 21.

Table 54: LD 12

Prompt	Response	Description
REQ	CHG	Change
TYPE	2250	Attendant console type
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx SCC yyyy	Speed Call Controller, where: xx = key number, and yyyy = list number (0-8190)

Feature operation

To store Speed Call entries from a Meridian 1 proprietary telephone, or attendant console (Controller):

- Without lifting the handset, press Speed Call. The indicator flashes.
- Dial the Speed Call code (0-999), followed by the telephone number it represents.
- Press Speed Call. If the entry is accepted, the indicator goes off. If the entry is not accepted, the indicator continues flashing.

To make a Speed Call from a Meridian 1 proprietary telephone, or attendant console (User):

- Lift the handset and press Speed Call (telephone).

  - Select an idle loop key and press Speed Call (attendant console).
- Dial the Speed Call code. The telephone number represented by the Speed Call code is dialed automatically.

To store Speed Call entries from an analog (500/2500-type) telephone (Controller):

- Lift the handset and press octothorpe (#) +2 (2500 telephone) or SPRE+75 (analog (500/2500-type) telephone).
- Dial the Speed Call code (0-999), followed by the telephone number it represents. If the entry is accepted, you hear silence. If the entry is not accepted, you hear a fast busy tone.
- Hang up.

Repeat steps 1 through 3 for each entry to be stored.

To make a Speed Call from an analog (500/2500-type) telephone (User):

- Lift the handset and dial #3 (2500 telephone), or SPRE 76 (analog (500/2500-type) telephone).
- Dial the Speed Call code (0-999). The telephone number represented by the Speed Call code is dialed automatically.

  In addition to SPRE codes your system may be equipped with Flexible Feature Codes (FFCs).

Chapter 28: Speed Call, System

Contents

This section contains information on the following topics:

Feature description

System Speed Call extends the capabilities of Speed Call. In addition to abbreviated dialing, System Speed Call allows a user to temporarily override the telephone's Class of Service, Trunk Group Access Restrictions (TGARs), and code restrictions.

Analog (500/2500-type) telephones, Meridian 1 proprietary telephones, and attendant consoles can activate System Speed Call by using SPRE or Flexible Feature Codes (FFC).

An analog (500/2500-type) telephone can be designated as a System Speed Call User only (not Controller) and can access one System Speed Call list. Meridian 1 proprietary telephones can be System Speed Call Users (SPRE codes or key access) or Controllers (key access only). Attendant consoles can be System Speed Call Users (dial access only) and System Speed Call Controllers (key access only).

Operating parameters

Up to 8191 (0-8190) Speed Call lists are allowed as long as sufficient memory is available. The new maximum includes all combined Speed Call, System Speed Call and Hot Line lists, 4096 (0-4095) of which can be System Speed Call lists.

System Speed Call lists can have up to 1000 entries and each entry can be up to 31 digits in length.

Restrictions applied to a telephone are ignored only for the origination of a call made through System Speed Call. Restrictions are applied if any call modification is attempted once the call is established.

System Speed Call lists can only be programmed in LD 18 or from telephones or attendant consoles equipped with a System Speed Call Controller key.

The technician can add or copy up to 100 System and regular Speed Call Lists at a time.

Feature interactions

Attendant Administration

System Speed Call lists can be assigned using Attendant Administration.

Authorization Code Security Enhancement

If the Basic Authorization Code (BAUT) or Network Authorization Code (NAUT) package is equipped, a Network Class of Service (NCOS) is assigned to the System Speed Call list. The NCOS of the System Speed Call list replaces the NCOS of the Authorization code or Forced Charge Account code if it increases the Facility Restriction Level (FRL) of the code.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using System Speed Call (SSU/SSC).

Basic/Network Alternate Route Selection (BARS/NARS)

If the BARS or NARS package is equipped, an NCOS is assigned to the System Speed Call list. The NCOS associated with the System Speed Call list replaces the NCOS of the telephone if it increases the Facility Restriction Level (FRL) of the user.

Calling Party Privacy

An outgoing trunk call initiated by dialing the Speed Call code will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored in the Speed Call Entry represented by the Speed Call code. The CPP code will be counted against the maximum number of digits (currently 31) allowed per Speed Call list entry.

A user can also store the CPP code in the Speed Call Entry (or Speed Call key). An outgoing CPP call can then be initiated by dialing the Speed Call code (or pressing the Speed Call key), followed by manually dialing the digits.

However, existing Speed Call limitations do not allow a user to dial *67 (or anything else) before accessing a Speed Call list entry.

Capacity Expansion

Any number from 0 to 4095 can be assigned to a System Speed Call list.

China - Flexible Feature Codes - Outgoing Call Barring

Digits dialed using System Speed Call are checked against the active OCB level.

Flexible Feature Code

With Flexible Feature Code (FFC), a confirmation tone is provided for Speed Call store after the end-of-dial (EOD) string is entered.

Hot Line

When the System Speed Call package is equipped, Hot Line lists have the characteristics and limitations of SSC lists. If the package is not equipped, Hot Line lists function like standard Speed Call lists.

Last Number Redial

A number dialed using a System Speed Call key becomes the Last Number Redial number on all telephones, except the M2317. A number dialed using SPRE-activated System Speed Call becomes the Last Number Redial number on all telephones. The original Class of Service and NCOS restrictions of the telephone apply when using Last Number Redial.

Off-Hook Alarm Security

Off-Hook Alarm Security (OHAS) treatment can apply to these features if the ASTM expires. The Alarm Security Timer may expire for the following reasons:

- A dial tone or interdigit timeout occurs while dialing the speed call access code.
- The Speed Call being accessed has an asterisk (*) causing a three-second delay. If the ASTM is three seconds or less, the OHAS intercept treatment may occur.

Pretranslation

Program a Speed Call List number to allow for Pretranslation. For example, if 9 pretranslates to 99 and you want to reach 99 nxx xxxx, you need to program the number in the Speed Call List as 9 nxx xxxx. When the Speed Call List is used, 9 nxx xxxx is pretranslated at call processing time to become 99 nxx xxxx.

Feature packaging

System Speed Call (SSC) package 34 has no feature package dependencies.

Feature implementation

Task summary list

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

1. Table 55: LD 17 on page 217

Set maximum number of Speed Call lists.

2. Table 56: LD 18 on page 217

   Compute Speed Call list memory size and disk records.
3. Table 57: LD 18 on page 218

   Add or change a System Speed Call list.
4. Table 58: LD 10 on page 218

   Add or change System Speed Call for Analog (500/2500-type) telephones.
5. Table 59: LD 11 on page 219

   Add or change System Speed Call list for Meridian 1 proprietary telephones.
6. Table 60: LD 12 on page 219

   Add or change a System Speed Call list for attendant consoles.
7. Table 61: LD 20 on page 219

   Print Speed Call data.

Table 55: LD 17

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PARM	Configuration Record. System parameters
...		
- MSCL	0-8190	Maximum number of Speed Call lists.

Table 56: LD 18

Prompt	Response	Description
Use this prompt sequence to determine if there is enough memory and disk space for new Speed Call lists. Compare the output with the MEM AVAIL and DISK AVAIL values output before the REQ prompt.		
REQ	COMP	Compute disk and memory.
TYPE	SCL	Speed Call lists.
NOLS	1-8191	Number of lists to be added.
DNSZ	4-31	Maximum length of DN allowed for Speed Call list.
SIZE	1-1000	Maximum number of entries in Speed Call list.

Table 57: LD 18

Prompt	Response	Description
REQ	NEW CHG OUT NEW xx, CPY xx	Add, change, or remove a single speed call list; Add or copy xx lists.
TYPE	SSC SCL	System Speed Call. Speed Call List.
LSNO	0-8190 xxxx yyyy	Number of list to add, where: xxxx = number of list to be copied, and yyyy = number of list to receive copy.
NCOS	0-99	NCOS to be assigned to calls accessing the list.
DNSZ	4-(16)-31	Maximum number of digits in a list entry (that is, 4, 8, 12, 16, 20, 24, 28, or 31).
SIZE	1-1000	Maximum number of entries in the Speed Call list.
WRT	(YES) NO	Data is correct and list may be updated.
STOR	xxx yy...yy	xxx = list entry number (0-9, 0-99, or 0-999). yy = digits to be stored against the entry (must be equal to or less than DNSZ).
WRT	(YES) NO	Data is correct and list may be updated.
The prompt WRT follows prompts SIZE and STOR asking you to confirm the correctness of the data just entered. If data is correct, enter YES or <CR>. A response of NO after the SIZE prompt causes all data entered to be ignored. A response of NO after the STOR prompt generates a warning message (SCH3213) indicating the data was not stored and must be reentered. A response of "****" aborts the program. Only the last STOR value is lost. All previous values to which WRT was YES are saved. The following information is output with the WRT prompt, following SIZE: ADDS: MEM: xxxxx DISK: yy.y Where xxxxx is the amount of protected memory and yy.y is the number of disk records required for the new Speed Call list. Check the MEM AVAIL and DISK REC AVAIL values output before the REQ prompt.		

Table 58: 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.
FTR	SSU yyyy	System Speed Call user, list number (0-4095).

Table 59: 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.
SSU	yyyy	System Speed Call list number (0-4095) for dial access.
KEY	xx SSU yyyy xx SSC yyyy	System Speed Call user key. System Speed Call Controller key, where: xx = key number, and yyyy = System Speed Call list number (0-4095). The M2317 must use key 21.

Table 60: LD 12

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SSU	yyyy	System Speed Call list number (0-4095) for dial access.
KEY	xx SSC yyyy	System Speed Call Controller key, where: xx = key number, and yyyy = System Speed Call list number (0-4095).

Table 61: LD 20

Prompt	Response	Description
Respond to the TYPE prompt with SCL to print regular and System Speed Call lists and pretranslation. Respond to the TYPE prompt with SSL to print the System Speed Call data block.		
REQ	PRT	Print.
TYPE	SCL	Regular and system speed call lists.
LSNO	0-8190	List number for speed call or system speed call print for all lists.
RNGE	xxxx xxxx	Range of all speed call entries (0-1000) to be printed. Print all entries

Feature operation

To store System Speed Call entries from a Meridian 1 proprietary telephone, or attendant console (Controller):

1. Without lifting the handset, press Speed Call. The indicator flashes.
2. Dial the Speed Call code (0-999), followed by the telephone number it represents.
3. Press Speed Call. If the entry is accepted, the indicator goes off. If the entry is not accepted, the indicator remains flashing.

To make a System Speed Call from a Meridian 1 proprietary telephone, or attendant console (User):

1. Lift the handset and dial SPRE 73 or press the System Speed Call key (telephone).

   –OR –

   Select an idle loop key and dial SPRE 73 (attendant console).
2. Dial the Speed Call code.

   If the Speed Call number is accepted, the telephone number represented by the Speed Call code is dialed automatically. No confirmation tone is given unless Flexible Feature Code (FFC) is implemented.

   If the Speed Call number is not accepted, a fast busy signal indicates the number was rejected.

To make a System Speed Call from an analog (500/2500-type) telephone (User):

1. Lift the handset and dial SPRE 73.
2. Dial the Speed Call code (0-999). The telephone number represented by the Speed Call code is dialed automatically.

   In addition to SPRE codes your system can be equipped with Flexible Feature Codes.

The routine to add a call list aborts under the following conditions:

- trying to add a call list whose number is already in use, or
- trying to add multiple call lists when there is insufficient memory.

Chapter 29: Speed Call or Autodial with Authorization Codes

Contents

This section contains information on the following topics:

Feature description

This feature is an enhancement to the existing Speed Call and Autodial features. It allows a Speed Call entry to contain an Authorization Code with an associated trunk route or Electronic Switched Network (ESN) access code and dialed number. The digits stored are recorded in Call Detail Recording (CDR), if equipped, for billing purposes.

The Speed Call entry can be one of the following:

- SPRE + 6 + Authorization Code
- SPRE + 6 + Authorization Code + #, or
- SPRE + 6 + Authorization Code + # + ESN access code and dialed number.

Operating parameters

Authorization Code Conditionally Last is not supported.

An octothorpe (#) is required as a delimiter after the Authorization Code if an ESN access code and dialed number are stored as part of the Speed Call or Autodial key. If the octothorpe is not entered, the user receives a fast busy tone. The octothorpe is not stored in the CDR record.

If the system initializes before the Authorization Code is recorded by CDR, the record may be lost.

An M2317 telephone can display up to 31 digits.

For Meridian 1 proprietary telephones, up to 31 digits per Speed Call entry are allowed.

On digit display sets, Authorization Codes cannot be blocked from being displayed.

There is no validation of the Authorization Code until the Speed Call key is activated.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The following packages are required to implement this feature:

- Basic Authorization Code (BAUT) package 25, or Network Authorization Code (NAUT) package 63.
- Optional features (OPTF) package 1, System Speed Call (SSC) package 34, or Network Speed Call (NSC) package 39.

Feature implementation

An Authorization Code can be entered as part of a Speed Call list.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 30: Station Activity Records

Contents

This section contains information on the following topics:

Feature description

When a telephone is configured with Class of Service Call Detail Monitoring Allowed (CDMA) for all incoming and outgoing calls, Station Activity Records are produced. The format of Station Activity Records is identical to other Call Detail Recording (CDR) records, but they have a new type of identifier (D). Existing CDR records are not affected by this new functionality.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Call Redirection

A Station Activity Record is only produced for a telephone designated as CDMA that is involved in a call with a trunk. A Station Activity Record is not generated for any telephone which does not answer the call, regardless of whether it has Class of Service CDMA or CDMD. Any other CDR records generated during call redirection are not affected.

Call Transfer

A Station Activity Record is generated when a telephone with Class of Service CDMA transfers a trunk call. CDR "X" record generation is not affected by this development. The telephone to which the call is transferred also produces a Station Activity Record if it has Class of Service CDMA and answers the call. When the second "D" record is produced (by the telephone to which the call is transferred), the digits field of the "D" record shows the digits dialed by the transferring telephone.

Conference

For a telephone with Class of Service CDMA involved in a call with a trunk, a Station Activity Record is produced only when that telephone conferences in the first party. Conferencing of all subsequent parties does not generate a "D" record. An additional "D" record is produced when the last conferee with Class of Service CDMA connected to the trunk goes on hook. This does not affect any other CDR record generation during a conference.

Internal Call Detail Recording

Internal Call Detail Recording records are produced according to the Class of Service ICDA/ ICDD of a telephone. The Station Activity Record enhancement does not affect the ICDR record generation.

Feature packaging

Station Activity Records is package 251 (SCDR).

Dependencies:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Terminal (CTY) package 5

Feature implementation

Task summary list

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

1. Table 62: LD 10 on page 225

   Set Class of Service CDMA/CDMD for an analog (500/2500-type) telephone.
2. Table 63: LD 11 on page 226

   Set Class of Service CDMA/CDMD for Meridian 1 proprietary telephones.
3. Table 64: LD 27 on page 226

   Set Class of Service CDMA/CDMD for BRI sets.
4. Table 65: LD 15 on page 226

   CDR must be enabled for the customer.

Table 62: LD 10

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	500	Analog (500/2500-type) telephone.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the telephone (when the trunk is involved in the call). CDMD denies record generation.

Table 63: LD 11

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the telephone (when the trunk is involved in the call). CDMD denies record generation.

Table 64: LD 27

Prompt	Response	Description
REQ	NEW CHG PRT	New, change, or print.
TYPE	DSL	Digital Subscriber Loop.
...		
CLS	(CDMD) CDMA	CDMA allows Station Activity Records to be generated for the telephone (when the trunk is involved in the call). CDMD denies record generation.

Table 65: LD 15

Prompt	Response	Description
REQ:	NEW CHG	New, or change.
TYPE:	CDR	Call Detail Recording
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- CDR	YES	Call Detail Recording.
- PORT	0-15	The CDR port number for the customer.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 31: Station Category Indication

Contents

This section contains information on the following topics:

Feature description

The Station Category Indication (SCI) feature allows an attendant to selectively answer internal attendant Directory Number (DN) calls on a priority basis. Stations are assigned a category, with priority indicated by an Incoming Call Indicator (ICI) lamp at each attendant console. Using the answering priority defined in LD 15, the attendant gives prompt attention to a call presented at a high-priority ICI lamp by selecting the associated ICI key.

Operating parameters

A maximum of seven station categories (1-7) can be assigned.

Calls from SCI 0 stations appear on the dial 0 ICI.

Calls from fully restricted stations appear on the dial 0 fully restricted ICI.

The Station Category Indication (SCI) feature should not be mixed with any other Incoming Call Indicator (ICI) assignment on the same ICI key/lamp pair.

Feature interactions

Centralized Attendant Service

When Centralized Attendant Service (CAS) is active, calls from a remote station to the attendant DN appear on the remote ICI key/lamp pair at the CAS main, regardless of the station SCI category.

Controlled Class of Service

The Controlled Class of Service (CCOS) feature has priority over SCI. A station's SCI category is suppressed when CCOS is active, and calls to the attendant DN carry the CCOS class defined in the database.

Phantom Terminal Numbers (TNs)

SCI cannot be enabled on a Phantom TN.

Feature packaging

Station Category Indication (SCI) package 80 has no feature package dependencies.

Feature implementation

Task summary list

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

1. Table 66: LD 15 on page 229

Add or change a Station Category Indication ICI key/lamp pair for attendant consoles.

2. Table 67: LD 10 on page 229

   - Change SCI for analog (500/2500-type) telephones.

3. Table 68: LD 11 on page 229

   Change SCI for Meridian 1 proprietary telephones.

Table 66: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	CDB	Customer Data Block.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
ICI	0-19 CA1-CA7	Assign ICI key/lamp pair for SCI.
ICI	0-19 DL0	Dial 0 (calls from telephones in SCI 0).
ICI	0-19 DFO	Fully restricted (call from fully restricted telephones).

Table 67: 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.
SCI	0-7	SCI number.

Table 68: 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.
SCI	0-7	SCI number.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 32: Station Specific Authorization Code

Contents

This section contains information on the following topics:

Feature description

Station Specific Authorization Code (SSAU) enables the system administrator to control the level of authorization code access on a per telephone basis. SSAU applies to analog (500/2500-type) telephones and Meridian 1 proprietary telephones; it does not apply to Basic Rate Interface (BRI) telephones.

Station Specific Authorization Code provides three levels of authorization code access:

- Authcode Unrestricted (AUTU)

  An AUTU telephone has no authorization code access limitations. Any authorization code is accepted and processed normally.

- Authcode Restricted (AUTR)

  An AUTR telephone can enter up to six assigned authorization codes. The authorization code entered must match one of the preassigned codes. Any other authorization code will be rejected and the call will not be completed.

- Authcode Denied (AUTD)

An AUTD telephone has no access to authorization codes. Any authorization code will be rejected and the call will not be completed.

Operating parameters

The same authorization code can be assigned to more than one AUTR telephone.

There is cross-checking between LDs 10 and 11, which define a station specific authorization code, and LD 88, which ensures that the user has entered a valid authorization code.

LD 88, which is used to delete an existing authorization code, does not check if the authorization code is assigned as a station specific authorization code before the deletion.

The Station Specific Authorization Code feature does not apply when the authorization code is prompted from a tandem node.

Feature interactions

Attendant Administration

Station Specific Authorization Code does not support Attendant Administration.

Authorization Code Security Enhancement

Users cannot freely enter authorization codes from telephones that have AUTR or AUTD Class of Service.

Autodial, Speed Call

The SSAU feature treats stored autodial numbers as if they were entered at the telephone.

Feature packaging

Station Specific Authorization Code (SSAU) is package 229, which requires Basic Authorization Codes (BAUT) package 25.

Feature implementation

Task summary list

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

1. Table 69: LD 88 on page 233

   Create Authorization Code data block.
2. Table 70: LD 88 on page 234

   Create an Authorization Code Table.
3. Table 71: LD 10 and LD 11 on page 234

   Activate SSAU. Use LD 10 or LD 11 according to telephone type.
4. Table 72: LD 20 on page 235

   Set Security Password.

Table 69: LD 88

Prompt	Response	Description
REQ	NEW	Create.
TYPE	AUB	Authcode data block.
CUST	xx	Customer number, as defined in LD 15.
SPWD	xxxx	Secure data password.
ALEN	1-14	Number of digits in authcodes.
ACDR	YES NO	Activate CDR for authcodes. There is no default.
RANR		RAN route number for "Authcode Last" prompt (NAUT).
	0-511	Range for Large System and CS 1000E system.

Prompt	Response	Description
CLAS	(0)-115	Class code value assigned to authcode (NAUT).
COS	aaa	Class of Service.
TGAR	(0)-31	Trunk Group Access Restrictions.
NCOS	(0)-99	Network Class of Service.
AUTO	YES NO	Automatically generate authcodes.
- SECR	0-9999	Security password (NAUT).
- NMBR	1-9999	Number of authcodes to be generated.
- CLAS	(0)-115	Class code value assigned to authcode (NAUT).

Table 70: LD 88

Prompt	Response	Description
REQ	NEW	Create.
TYPE	AUT	Authorization Code Table.
CUST	xx	Customer number, as defined in LD 15.
SPWD	xxxx	Secure data password.
CODE	xxxx	Authcode (number of digits must equal ALEN).
CLAS	(0)-115	Class code value assigned to authcode (NAUT).

Table 71: LD 10 and LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add, or modify.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
CLS	(AUTU) AUTR AUTD	Authcode unrestricted. Authcode restricted. Authcode denied.
MAUT	(NO) YES	Modify assigned authcodes for this telephone.
SPWD	xxxx	Correct security password (if one is defined).
AUTH	x nnnn	x is in the range of 1-6; nnnn is the assigned authcode (a valid authorization code defined in LD 88).
	X x	X x deletes an assigned authcode.
Changing an AUTR telephone to AUTU or AUTD clears all assigned authcode information previously defined for that telephone.		

Table 72: LD 20

Prompt	Response	Description
REQ:	PRT	Print.
TYPE:	xxx	Type of 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.
CDEN	SD DD 4D 8D	Card density requested.
CUST	xx	Customer number, as defined in LD 15.
SPWD	xxxx	Valid Security data password to display SSAU.
Once SPWD is prompted, a valid security data password as defined in the customer data block is required for displaying Authorization (AUTH) information for sets with Class of Service Authorization Code. Sets with Class of Service Authcode Unrestricted (AUTU) and Authcode Denied (AUTD) do not have AUTH information for display. Entering of a carriage return at the SPWD prompt will result in the AUTH information being skipped during printing.		

In LD 20, Security Password (SPWD) will not be prompted if any of the following conditions exists:

- the Station Specific Authcode Package 220 is not equipped,
- the response to the TN prompt is more than one specific TN,
- the response to the TN prompt is a unique TN, but the customer of this TN does not have a security data password defined,
- the response to the CUST prompt is not a specific customer, or
- the response to the CUST prompt is a specific customer number but the customer does not have a security password defined.

Feature operation

After an authorization code is entered, the Station Specific Authorization Code feature determines if the telephone is allowed to use the entered code. If the authorization code is not allowed on that telephone, the existing invalid authorization code treatment occurs. Otherwise, normal authorization code processing occurs.

Chapter 33: Station-to-Station Calling

Contents

This section contains information on the following topics:

Feature description

Station-to-Station Calling allows direct dialing between station users in the same customer group without the assistance of the attendant.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Manual Line Service

If a single line telephone has been assigned a Manual Line Class of Service, the telephone automatically rings the attendant when it goes off-hook.

Private Lines

You must go over the public network to reach a Private Line. The software PRDN is not meant to be dialed directly.

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 34: Stored Number Redial

Contents

This section contains information on the following topics:

Feature description

Stored Number Redial (SNR) allows telephones and attendant consoles to store one previously dialed number of 4 to 31 digits for automatic redialing.

Depending on the type of telephone, the number can be stored before a call is placed, during Ringback, while the number is busy, or during an active call. On attendant consoles, the number can be stored only before a call is placed. Stored Number Redial (SNR) is not supported on M2317 telephones or analog (500/2500-type) telephones serving as Private Lines.

Operating parameters

When a number is stored, it overwrites any previously stored number.

Storage is limited to one number per analog (500/2500-type) telephone and one number per SNR key. When a call is established through a Tandem TIE Trunk Network (TTTN), the user is required to pause for dial tone. When you store a number using SNR, automatic redialing may fail because required delays are not added. It is possible to include delays in the outpulsing by dialing the asterisk (*) in the original digit string where dial tone is expected. Each asterisk (*) signifies a three-second delay in outpulsing.

The asterisk (*) used to introduce a pause while outpulsing digits is supported on analog and DTI trunks, but not supported on ISDN trunks. On ISDN trunks, if the OPAO feature is enabled, the asterisk (*) is outpulsed as a called party digit.

The three-second delay is not available from a 500-type telephone.

During the stored Number Redial (SNR) programming mode, if the user attempts to store more digits than the maximum number defined for the telephone or console, SNR programming is canceled and overflow tone is returned. During an active call on a Meridian 1 proprietary telephone, if a user attempts to store more digits than the specified limit, the SNR operation fails, the previously stored number remains unchanged, and a failure indication is not given. The SNR indicator remains off.

For analog (500/2500-type) telephones, in order to store a number dialed to a busy DN, the maximum length of the stored number must be at least five digits (see prompt FTR RDL xx in LD 10).

Feature interactions

Authorization Code Security Enhancement, Charge Account, Forced Charge Account

The Authorization, Charge Account, and Forced Charge Account codes are not stored. To store a code, dial the code prior to using Stored Number Redial to dial the call.

Automatic Redial

The Automatic Redial (ARDL) feature can be activated on a call using the Stored Number Redial (RDL) key.

Calling Party Privacy

During Stored Number Redial (SNR) programming, a user can store the Calling Party Privacy (CPP) code followed by the normal dialing sequence in the SNR data space. Outgoing calls originated by the SNR feature will send the Privacy Indicator to the far end. The CPP code will be counted against the maximum number of digits (currently 31) allowed by the SNR feature.

During an active call on a Meridian 1 proprietary telephone, the Stored Number Redial feature will store the CPP code in the SNR data space if the CPP code was included in the number dialed by the originator. The outgoing redialed calls will send the Privacy Indicator to the far end.

China Number 1 Signaling Enhancements

Delay Digit Outpulsing will be denied when dialing is done by way of Stored Number Redial.

End-to-End Signaling

End-to-End Signaling (EES) activates after a call to a trunk is established by expiration of the end-of-dial timer. Further digits dialed are not stored by the SNR feature once it is in EES mode.

Group Hunt

A Pilot DN will be stored as a Stored Number Redial (SNR) number when it is dialed directly.

Intercept Computer Dial from Directory - Post-dial Operation

An attendant can dial an extension from the Intercept Computer, and then press the Stored Number Redial key to store the called number (following the rules of the Stored Number Redial feature).

Multi-Party Operations

For analog (500/2500-type) telephones, the Last Number Redial/Stored Number Redial feature can be used when normal or special dial tone is received. The last number redialed that can be stored is the first call of a consultation connection, and can be stored only after the connection is completely released.

Feature packaging

Stored Number Redial (SNR) package 64 has no feature package dependencies.

Feature implementation

Task summary list

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

1. Table 73: LD 10 on page 242

   Add or change SNR for Analog (500/2500-type) telephones.
2. Table 74: LD 11 on page 242

   Add or change SNR for Meridian 1 proprietary telephones.
3. Table 75: LD 12 on page 243

   Add or change SNR for attendant consoles.

Table 73: 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	(XFD) XFA	Call Transfer (denied) allowed.
FTR	RDL xx	Activate SNR, where: xx = the maximum number of digits that can be stored (that is, 4, 8, 12, (16), 20, 24, 28, 31).

Table 74: 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.
KEY	xx RDL yy	Add an SNR key, where

Prompt	Response	Description
		xx = key number, and yy = the maximum number of digits that can be stored (that is, 4, 8, 12, (16), 20, 24, 28, 31).

Table 75: LD 12

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx RDL	Add an SNR key.

Feature operation

Attendant consoles, Meridian 1 proprietary telephones

To store a number prior to dialing (for attendant consoles, and Meridian 1 proprietary telephones):

Without lifting the handset, press Stored No.

- Dial the number.
- Press Stored No. again. The number is stored, replacing any previous one.

To store a number during Ringback, while the number is busy, or during an active call (for Meridian 1 proprietary telephones only):

- Press Stored No.

To call a stored number:

- Press DN (Meridian 1 proprietary telephones) or the Loop key (consoles).
- Press Stored No. The number is dialed.

Analog (500/2500-type) telephones

To store a number prior to dialing:

- Lift the handset.
- Dial SPRE 78, or the Flexible Feature Code (FFC) assigned for SNR.
- Dial the number to be stored.
- Hang up. The number is stored, replacing any previous one.

To store a number before a call is placed, during Ringback, while the number is busy, or during an active call:

- Flash the switchhook or press LINK.
- Dial SPRE 78, or the FFC assigned for SNR.

To call a stored number:

- Lift the handset.
- Dial SPRE 79, or the FFC assigned for SNR. The number is dialed.

Chapter 35: Supervised Analog Lines

Contents

This section contains information on the following topics:

Feature description

The Supervised Analog Lines feature provides two types of call supervision signaling capabilities: battery reversal answer/disconnect supervision; and hook flash disconnect supervision. These forms of supervision are provided to terminal devices connected to analog ports in the system.

Battery Reversal Supervision

Battery reversal answer and disconnect supervision signaling is used for calls originating from the terminal device. It provides both far-end (that is, the called party) answer supervision and far-end disconnect supervision signals to the terminal device. It does not apply to incoming calls terminating at the terminal device.

In the idle state, the analog port in the system provides ground signal on the tip lead and battery on the ring lead. This polarity is maintained during dialing and ringing at the far end. When the far end answers, the battery and ground connections are reversed. The reverse battery is maintained while the call is established. When the far end disconnects, the battery and ground connections are reverted to the idle state to signal that the far end has disconnected. If the terminal device disconnects first, the system sends the Deactivate Battery Reversal Scan

Signal Distribution (SSD) message to the firmware after receiving the on-hook status to revert the polarity to its idle state.

Two types of battery reversal are supported. Battery Reversal for Absolute Answer Only provides an answer supervision signal to the terminal device only when the system detects an absolute answer. Battery Reversal for Absolute and Assumed Answer provides an answer supervision signal to the terminal device even when an assumed answer is detected and the far end is not capable of indicating definite answer (for example, an outgoing call on an unsupervised loop start trunk).

Hook Flash Disconnect Supervision

Hook flash disconnect supervision is used for incoming calls terminating at the terminal device. The disconnect signal is indicated by the removal of the ground connection to the tip lead for a specific period of time, which is provided by firmware ranging from a minimum of 10 milliseconds to a maximum of 2.55 seconds. The analog port is held busy for incoming calls while hook flash is in progress.

Operating parameters

This feature applies to Intelligent Peripheral Equipment that support the Supervised Analog Line feature only.

Supervised Analog Lines require NT1R20AB off premise line cards. However, NT5D11AA or NT5D14AA line cards may also be used.

Disconnect supervision is not provided to the terminal device if the system does not receive any indication of the far end releasing.

If the system does not receive any answer indication, and answer supervision is not extended to the terminal device following an assumed answer condition, disconnect supervision cannot be extended when the far end disconnects.

If the Battery Reversal Supervision feature is configured for an analog line on an analog card that does not support battery reversal, the battery reversal SSD messages from the system software are ignored by the analog card firmware. In this case, no battery reversal signal is extended to the terminal device.

If the Hook Flash Disconnect Supervision feature is configured for an analog line on an analog card that does not support hook flash, the hook flash SSD messages from the system software are ignored by the analog line card firmware. In this case, no hook flash signal is extended to the terminal device.

If the system initializes while an outgoing call originating from an analog line is established and battery reversal is activated, unprotected data is lost. In this case, battery reversal remains activated when the call is cleared down by either party. However, the line status is reverted to normal when the next outgoing call is answered and then cleared down.

If the hook flash timer is set equal to or greater than the on-hook timer, activation of the hook flash disconnect signal also causes the card to send an on-hook message and then an off-hook message to the system. In this case, if the user remains off-hook after the far end disconnects, dial tone is received and an outgoing call can be initiated.

Feature interactions

Call Transfer

If more than one active call is extended to an analog line, the call type associated with an analog line is determined by the first active call. The call type is assumed to be incoming and hook flash supervision applies if a terminal device answers an incoming call from an idle state. If the terminal device performs a switch hook flash to put the first party on hold and initiates a consultation call, the Battery Reversal feature is not supported; no battery reversal answer signal is extended to the terminal device when the second party answers.

If the first party disconnects while the terminal device is connected to the second party, no disconnect supervision is extended to the terminal device. However, hook flash disconnect supervision is extended to the terminal device when the second party disconnects (that is, a disconnect supervision signal is sent only when the last party connected to the terminal disconnects).

If a terminal device originates an outgoing call, battery reversal answer supervision is extended when the called party answers. The polarity of the line remains reversed when the terminal device performs a switch hook flash and then initiates a consultation call to a second party. The analog line is reverted to normal polarity when the terminal device completes the transfer and drops out or when the last of either the held party or the consultation party disconnects.

Conference

If a terminal device answers an incoming call and then initiates a conference, no battery reversal answer supervision signal is extended to the terminal device when new parties of the conference answer. However, a hook flash disconnect supervision signal is extended to the terminal device when the last party in the conference disconnects.

If a terminal device initiates a conference, battery reversal answer supervision is extended to the terminal device when the first party answers. No polarity change is made when additional parties are added to the conference. The polarity is reverted to normal when the terminal device disconnects or when the last party in the conference disconnects.

Multi-Party Operations

As in the cases with Call Transfer and Conference, the call type of the first active call determines whether battery reversal or hook flash supervision applies. Also, supervision signaling is not supported for the second call. A disconnect supervision signal is extended only when the last party disconnects.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 76: LD 10 on page 248

   Enable battery reversal supervision.
2. Table 77: LD 10 on page 249

   Enable hook flash disconnect supervision.

Table 76: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add, or 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.
...		

Prompt	Response	Description
FTR	OSP (1)	Outgoing call supervision. Answer and disconnect supervision for outgoing calls with absolute and assumed answer indication. If the numeric parameter is not entered and the saved value is null, it is defaulted to 1. Otherwise it remains unchanged.
	OSP 2	Answer and disconnect supervision for outgoing calls with absolute answer supervision only.
	XOSP	Enter XOSP to disable battery reversal answer and disconnect supervision.

Table 77: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add, or 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.
...		
FTR	ISP 1...(75)...255	Enable hook flash disconnect supervision with flash timer in 10 millisecond units. If the numeric parameter is not entered and the saved value is null, it is defaulted to 75. Otherwise it remains unchanged.
	XISP	Enter XISP to disable hook flash disconnect supervision.

Respond to the FTR prompt in LD 10 with OSP 1, and then with ISP 1...(75)...255 to enable both battery reversal supervision and hook flash disconnect supervision.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 36: Telelink Mobility Switch 1

Contents

This section contains information on the following topics:

Feature description

The Telelink Mobility Switch 1 feature allows a system in conjunction with the Mobility Control Point (MCP) application to deliver a call from the Public Switched Telephone Network (PSTN) to a portable telephone subscriber. The portable telephone subscriber does not need to have a telephone physically resident in the system. A unique personal Directory Number not related to a physical termination is assigned for each subscriber of a portable. This unique personal directory number is defined as a Dialed Number Identification Services (DNIS) number within the system switch. Digit conversion is used to translate the incoming DNIS number to a Controlled DN (CDN). The CDN contains the Application Module Link (AML) number that connects the system to the MCP application.

When a call comes to the system (acting as the mobility switch) from the PSTN via a DNIS Incoming Digit Conversion (IDC) trunk, the call is terminated to a CDN by the system software through IDC operation. The last few dialed digits are saved as a DNIS (subscriber identity) number.

A CDN can be operated in controlled or default mode. If in controlled mode, call treatment is controlled by the MCP application. If in default mode, call treatment is handled by the system software and default treatment is given to the call.

When a Personal Communications Service (PCS) call terminates to a CDN which is in the controlled mode, the system will notify the MCP application by providing the call's incoming route and DNIS (subscriber identify) number. This enables the MCP application to ascertain which subscriber the caller desires to reach.

The MCP application has a table containing the last zone in which each subscriber is registered, so that the MCP application can send a message to the correct zone to find out the idle/busy status of the portable telephone subscriber.

If a subscriber is busy or unable to answer the call, the MCP application will request the system (acting as a mobility switch) to either return busy or overflow tone to the caller. If the called party has subscribed to a voice messaging service, the MCP can request the system (acting as a mobility switch) to allow the caller to leave a voice message.

If the called party is idle, the MCP application will request the system (acting as a mobility switch) to optimally give ringback, provide a Recorded Announcement (RAN) or give silence to the caller, while the MCP application requests the system to make an outgoing call that will be used to alert the called party that an incoming call is waiting. This outgoing call is initiated from a phantom TN. The phantom TN does not need a physical line connection or telephone in the system. The phantom TN needs to be assigned as an associated telephone so that the MCP application will get status messages regarding the state of the phantom telephone. The public number of this outgoing call will be provided by a Zone Controller (ZC). The ZC reserves this incoming line which is connected to the public number.

When the phantom call is received on the reserved line, the ZC alerts the called party's portable. If MCP has requested silence for the call, then at this time the MCP will request ringback treatment for the call. Once the phantom call is answered by the subscriber, the ZC notifies the MCP application. Subsequently, the MCP application requests the system to merge the two related calls (an incoming call in the CDN queue and an outgoing call to the called party), so that the caller of the incoming call and the called party can speak to each other.

When an incoming call terminates to a CDN that is in default mode, the system (acting as a mobility switch) allows the caller to leave a voice message for the called party or give overflow tone to the caller when the call ceiling is exceeded. The CDN will be in default mode under abnormal conditions such as the AML, MCP or Application Programmable Interface (API) going down.

The Mobility Switch will also provide centralized voice prompts in lieu of zones if an exception condition is encountered when a portable is attempting to make an outgoing call.

Operating parameters

There will be a 3 DB loss on a DTI trunk when a Digital Trunk Interface (DTI) trunk is involved in a merge call, and a 0 DB loss on a Primary Rate Interface (PRI) trunk when a PRI trunk is involved in a merge call. It is therefore recommended that a PRI trunk should be used on the Mobility Switch instead of a DTI trunk.

Calls to subscribers without physical sets on the system must be originated from DNIS routes.

The MCP application should request Force Overflow to an incoming call when the DNIS information is not present in an AML-ICC message.

This feature is supported for North American markets only.

ACD-C or ACD-D reports are not a requirement of this feature. Operational measurements of PCS calls are supported by the MCP application.

External calls coming to a CDN with Value Added Server Identification (VASID) connected to the MCP application from a DID (Digital or ISDN) trunk will only be supported by this feature. For this reason disconnect supervision will be guaranteed to be returned to the system (acting as a mobility switch) when the call is disconnected.

This feature only supports TIE, CO ground start (analog, digital or ISDN) trunks as the outgoing trunk of a phantom call with ZC as the destination. For this reason, disconnect supervision must be obtained from the far end when the call is disconnected by the far end.

Combination of CDNs and ACD-DNs (Interactive Voice Response-DN) cannot exceed 240 per each customer on the system (acting as a mobility switch).

The maximum number of routes cannot exceed 512.

The maximum number of IDC/New Flexible Code Restriction (NFCR) Translation Tables per customer cannot exceed 255.

Due to the Federal Communications Commission (FCC) ruling, answer supervision is required to be returned if a Give Silence or Give Music is provided as a first call treatment to a PCS (incoming DID) call as per current operation.

If the system (acting as a mobility switch) initializes, all calls waiting in the CDN queues will be lost. The AML-INIT message will be sent to applications when an initialization occurs. When the MCP application receives an INIT message from the system (acting as a mobility switch), it erases information on existing calls.

A maximum of five Device Groups (DGRPs) will be supported per customer. An Associated Set (AST) Meridian 1 proprietary telephone with Idle Terminal for Third Party Application (ITNA) enabled can only be grouped to one DGRP.

The originator of an outgoing phantom call must be a phantom TN which is an AST Meridian 1 proprietary telephone with ITNA enabled.

An attendant telephone and a Basic Rate Interface (BRI) telephone will not be allowed to merge a call to another telephone or trunk.

When two trunks are joined, at least one trunk must have disconnect supervision.

The Application Module (AM Base) that interfaces with the MCP application cannot control more than one application (that is, MCP and Customer Controlled Routing applications are not supported).

A call that is initiated from the phantom telephone must be in established state, before it can be merged with the caller. If answer supervision is defined for the outgoing trunk, the call from the phantom telephone will be put in established state when the answer supervision answer is returned to the system (acting as a mobility switch). If answer supervision is not defined for the outgoing trunk, the call from the phantom telephone will be put in established state when the End-of-dialing timer has expired (128-32,640 msecs. after the last digit has been sent out).

If answer supervision is not defined for the outgoing trunk to which the phantom trunk is connected, it is possible that a random call may beat the phantom call to the reserved line and the caller will be given a busy tone.

If answer supervision is defined for the outgoing trunk, it is possible that the PSTN might not return the answer supervision signal to the system. If answer supervision is not returned, the system will not allow the phantom telephone call to be merged to the caller.

If answer supervision is defined for the outgoing trunk, it is possible that the answer supervision signal could be significantly delayed across the PSTN if the signal goes through many tandem Central Offices. This causes a subsequent delay between the time the subscriber answers the portable and the time when the incoming call is connected.

No Message Waiting Indication will be sent to the MCP application when a caller has left a voice message.

The MCP application must return a dialable number to the system (acting as a mobility switch) to launch the outgoing call to the Zone Controller. The dialable number includes the ESN access code if necessary.

If 1+ dialing is required at the first Central Office that the phantom call goes to from the system, it should either be provided by the MCP application or inserted via digit manipulation on the mobility switch.

Multi-purpose Serial Data Link (MSDL) (NT6D80AA) is required to connect the system to the AM or a host.

Feature interactions

The following features interact with the Telelink Mobility Switch 1 feature:

- Report Control
- Print CDN Parameters and Options Command
- CNTL Command (determines whether CDN is in controlled mode)
- DFDN Command (sets default of ACD-DN)
- CEIL Command (controls ceiling of the CDN)
- Supervisor Control of Queue Size

- Overflow by Count
- Attendant Extension
- Attendant Recall
- Network ACD (NACD)
- Timed Overflow and Enhanced Overflow
- Display Waiting Calls (DWC key)
- Night Service
- Transition Mode via the Night Service key
- Night Mode via the Night Service key
- Incoming Digit Conversion
- Night Key Digit Manipulation
- Call Forward No Answer
- Call Forward No Answer (Second Level)
- Call Forward All Calls
- Internal Call Forward
- Feature Invocation Messages
- Hunting
- Call Forward Busy
- Remote Call Forward
- Attendant and Network Wide Remote Call Forward
- Network Call Redirection
- Call Forward Override
- Trunk Optimization
- Call Transfer
- Call Transfer – By Interactive Voice Response Unit
- Conference
- Conference – By Interactive Voice Response Unit
- No Hold Conference
- Calling Line Identification
- Basic Rate Interface
- Incremental Software Management
- PBS Set Line Disconnect

- Application Module Link Enhancements
- NCOS Restrictions
- Time-of-day Routing
- Expensive Route Warning Tone
- Off-hook Queuing
- Call Back Queuing
- Remote Virtual Queuing
- Authcode Last
- Equal Access
- 1+ Dialing
- Interchangeable Numbering Plan Area
- Inter Digit Pretranslation
- Free Call Area Screening
- New Flexible Code Restriction
- Call Forward on DumpSysload
- Flexible Numbering Plan
- Multi-party Operation
- Priority Override
- Group Hunt
- Virtual Network Services
- Originator Routing Control, and
- Enhanced Night Service.

The following are a list of features that interact with the Merge Call aspect of this feature:

- Tenant-to-tenant Access
- Class of Service Restrictions
- Network Class of Service (NCOS) Restrictions
- Trunk Group Access Restrictions
- Schedule Access Restrictions
- Trunk Barring
- Feature Group D
- Attendant Barge-in
- Attendant Break-in

- Attendant Busy Verify
- Transfer
- Conference
- No Hold Conference
- Call Waiting
- Internal Call Waiting
- Group Call
- Voice Call
- Call Park
- Station Camp-on
- Dial Intercom
- ACD-DN key
- ACD Emergency/Answer Emergency keys
- ACD Call Agent/Answer Supervisor keys
- ACD Summon Supervisor/Answer Agent keys
- Single Call Arrangement DN keys
- Multiple Call Arrangement DN keys
- HOT Line
- Private Line
- Integrated Service Access Routes
- Integrated Signaling Link
- Application Module Link Unsolicited Status Message, and
- Application Module Link Call Abandoned Message, and Digit Display.

Feature packaging

This feature is included in base system software.

- Automatic Call Distribution, Package B (ACDB) package 41
- Network Alternate Route Selection (NARS) package 58
- Command Status Link (CSL) package 77
- Dialed Number Identification Services (DNIS) package 98
- Incoming DID Digit Conversion (IDC) package 113

- Application Module Link (AML) package 153
- Meridian Link Module (MLM) package 209
- Enhanced ACD Routing (EAR) package 214
- Enhanced Call Treatment (EACT) package 215
- Hold in Queue for Interactive Voice Response (IVR) package 218
- Call Identification (CID) package 247
- Phantom Terminal Number Operation (PHTN) package 254

The following packages are not required, but provide additional functionality:

- Call Detail Recording (CDR) package 4
- Office Data Administration System (ODAS) package 20
- ACD Load Management (LMAN) package 43
- Multi-user Login (MULI) package 242

Feature implementation

Task summary list

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

1. Table 78: LD 11 on page 258
2. Table 79: LD 17 on page 259

   Configure a phantom loop.
3. Table 80: LD 97 on page 260

   Create a phantom superloop.
4. Table 81: LD 23 on page 260

   Administer a service change to a CDN data block.

Table 78: LD 11

Prompt	Response	Description
REQ:	NEW CHG MOV OUT END CHG	New, change, move, out, end, or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.

Prompt	Response	Description
CUST	xx	Customer number, as defined in LD 15
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
		If l is a phantom loop and the CSL package is not equipped, an error message will be returned.
TOTN		Destination Terminal Number; prompted when REQ = MOV.
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CFTN		Copy From Terminal Number; prompted when REQ = CPY.
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SFMT	AUTO DN TN TNDN	DNs and TNs are assigned automatically. User enters the DN for each new telephone. User enters the TN for each new telephone. User enters DN and TN for each new telephone. Prompted when REQ = CPY.
CDEN	YES	Single, double or quad density (not prompted for superloop).
DES		ODAS designator.
...		
CLS	(NDD) (DNDD) ...	Class of service options. No digit displayed. Dialed name display denied. Block SPV and AGN if this TN is on a phantom loop.
AST	xx yy	Associate telephone assignment for Meridian Link application.
IAPG	(0)-15	Meridian Link Unsolicited Status Message (USM) group.
ITNA	(NO) YES	Idle TN for the third party application.
DGRP	(1)-5	Device group
...		
KEY	xx SCR yyyy	Single call ringing DN key.

Table 79: LD 17

Prompt	Response	Description
REQ	CHG END	Change, or end.

Prompt	Response	Description
TYPE	CFN CEQU	Configuration Record. Gate opener.
CEQU	YES	Change Common Equipment parameters. This will be prompted if TYPE = CFN.
...		
- TERM	0-159 0-159... [X] 0-159 [C] 0-159...	Single density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMO	0-159 0-159... [X] 0-159	Single density remote terminal loops. Precede loop number with X to remove.
- TERD	0-159 0-159... [X] 0-159 [C] 0-159...	Double density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMD	0-159 0-159... [X] 0-159	Double density terminal loops.
- TERQ	0-159 0-159... [X] 0-159 [C] 0-159...	Quad density local terminal loops. Precede loop number with X to remove. Precede the loop number with C to create a phantom loop.
- REMQ	0-159 0-159... [X] 0-159	Quad density remote terminal loops. Precede loop number with X to remove.

Table 80: LD 97

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SUPL	Superloop data.
SUPL	0-156 [X] 0-156 [C] 0-156	Superloop number in multiples of four. Precede loop number with X to remove a superloop. Precede the loop number with C to create a phantom superloop.

Table 81: LD 23

Prompt	Response	Description
REQ	NEW CHG	New, or change.
TYPE	CDN	Controlled DN data block.
CUST	xx	Customer number, as defined in LD 15
CDN	Directory Number	Controlled DN
FRRT	RAN route number	First RAN route number.

Prompt	Response	Description
FRT	1-2044	First RAN timer.
SRRT	RAN route number	Second RAN route number.
SRT	1-2044	Second RAN route timer.
FROA	(NO) YES	First RAN to be given immediately.
MURT	Music route	Music route number.
DFDN	Directory Number	Local default ACD-DN or IVR DN.
CEIL	0-(2047)	Call ceiling value.
OVFL	(NO) YES	Force Overflow Tone to the call when ceiling threshold exceeded?
TDNS	(NO) YES	Is the DNIS number an original party?
RPRT	(NO), YES	Information about this ACD-DN (or CDN) will be (excluded) included in management reports and status displays.
CNTL	(NO) YES	Is this CDN in controlled mode?
VSID	0-15	VASID for AML for application.
HSID	0-15	VASID for AML for host.
CWTH	0-(1)-2047	Call waiting LED threshold.
BYTH	(0)-2047	Busy queue threshold.
OVTH	0-(2047)	Overflow queue threshold.
STIO	1 2 3 ... 15	TTYs assigned for status displays.
TSFT	0-510	Telephone service factor threshold.
ACNT	xxxx	Default activity code.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 37: Telephones

Several different types of telephones are supported. Regular analog telephones are compatible with the system, as well as several special business telephones designed specifically to take advantage of the many features available on your system.

Ask your Avaya representative which telephone types are supported on your system.

For more information about telephones and consoles, see the following documents:

- *Avaya WLAN IP Telephony Installation and Commissioning, NN43001-504*
- *Avaya Attendant PC Console Fundamentals, NN43001-520*
- *Avaya Telephones and Consoles Fundamentals, NN43001-567*
- *Avaya IP Phones Fundamentals, NN43001-368*
- *Avaya DECT Fundamentals, NN43120-114*

Chapter 38: Teletype Terminal Access Control in Multi-customer Environment

Contents

This section contains information on the following topics:

Feature description

This is an enhancement of password usage for the Limited Access to Overlays feature. Under the previously enhanced operation, if no teletype terminal (TTY) activity had occurred for 20 minutes, the system automatically logged off. This value could not be changed. Counters were used to record the number of login attempts made on each TTY. If the threshold for the number of invalid attempts was exceeded, the system rejected any further activity at that port, for a defined period of time. No alarm mechanism was activated. Any attempt to log onto the system during this period of lockout was recorded by the system.

The prompt (LOUT) in LD 17 allows the TTY administrator (PWD2 user) to define a period of time (1-30 minutes) after which the system automatically logs off if no terminal activity has occurred.

The recording of invalid attempts remains the same as before. However, if the threshold for the number of invalid entries is reached, an alarm is activated; this alarm is in the form of the minor alarm lamp being lit on attendant consoles for all customers of the system. As was the case for the previously enhanced operation, an OVL400 message is sent to all active maintenance ports and to the first TTY administrator that logs on. Other treatments also remain the same.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Intercept Computer

The Intercept Computer (ICP) feature uses maintenance LD 51 to update the system with the intercept service interface information that it stored. This overlay logs off after five minutes if no messages have been received from the Intercept Computer. This five-minute period takes precedence over the value entered in response to the LOUT prompt in LD 17. If this value is less than five minutes, the system will wait for five minutes before logging off.

Feature packaging

International Supplementary Features (SUPP) package 131; Limited Access to Overlays (LAPW) package 164.

Feature implementation

Table 82: LD 17 - Configure TTY Access Control parameters.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN PWD	Configuration Record. Gate opener.
- NPW1	xxx	New Password 1

Prompt	Response	Description
- LOUT	1-(20)-30	Enter the time, in minutes, after which the system logs off if no terminal activity is detected.
...		
- FLTH	(0)-7	Enter the threshold for failed log-in attempts.
- LOCK	0-(60)-270	Enter, in minutes, the time the port is locked out once the FLTH has been reached.
- FLTA	(NO) YES	Enter YES to have the alarm activated once FLTH has been reached.
- AUDT	(NO) YES	Enter YES to have an audit trail activated for password usage.
- - SIZE	(50)-100	Prompted if AUDT = YES. Enter the size of the audit trail buffer.
- LLID	(NO) YES	Enter YES to activate the display of the last failed log-in attempt usage.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 39: Telset Call Timer Enhancement

Contents

This section contains information on the following topics:

Feature description

The Meridian digital telephones have displayable call timers, which start after the End-of-dialing (EOD) time out expires, and not when the called party answers. With this enhancement, the call timers on these telephones do not start until a true answer is detected on all trunks with answer supervision. These include the following:

- internal stations and attendants
- ground start and loop start supervisory trunks
- Direct Inward Dialing (DID) and Direct Outward Dialing (DOD) trunks
- Digital Trunk Interface (DTI) trunks
- Primary Rate Interface (PRI) trunks, and
- TIE trunks.

On trunks without answer supervision, the call timer starts at the EOD time out.

The feature operates in standalone or Integrated Services Digital Network (ISDN) environments.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 40: Three-Wire Analog Trunk - Commonwealth of Independent States

Contents

This section contains information on the following topics:

Feature description

The Three Wire Analog Trunk – Commonwealth of Independent States (CIS) feature provides the connectivity between the system and the three-wire analog trunks (3WT) used in the CIS. Analog incoming local three-wire trunks, analog incoming toll three-wire trunks, and analog outgoing Direct Inward Dialing (DID) three-wire trunks can be connected to the system.

Cards supported in an Intelligent Peripheral Equipment (IPE) environment are referred to as X3W cards. The following XW3 cards are supported by the Three-Wire Analog Trunk – CIS feature:

- NT5K60AA for incoming local and toll trunks
- NT5K61AA for outgoing trunks

The following functions are provided by the Three-Wire Analog Trunk – CIS feature:

- Delivery of Automatic Number Identification (ANI) on request from the Public Exchange/ Central Office for outgoing 3WT analog calls
- Downloading of specific transmission parameters (that is, pad data, public network toll access code, and hardware ID) for X3W cards, and
- Provision of dial tone internally by the system to the originator of the call after seizure of an outgoing X3W trunk.

The trunk state change validation timing is performed by the 3WT cards. For 3WT trunks, the originating party controls the disconnection of a call. When the originating party goes on-hook, the call is released. Note however, that when Malicious Call Trace is enabled, the Local Exchange may require a two-way release. This two-way release applies only on a telephone.

A 3WT Unproductive Timer is used to prevent a call on a X3W trunk from remaining unanswered for too long. This timer can be set to a maximum of 10 minutes.

For outgoing calls, digits are sent from the main Central Processing Unit (CPU) to the 3WT firmware. This is done by Dual-tone Multifrequency (DTMF) signaling for E3W equipment, and by IPE messaging for X3W equipment. The firmware then sends the digits as pulses and controls the actual decadic outpulsing.

Digits for incoming calls are received by the 3WT firmware as pulses. For E3W equipment, each valid pulse is reported to the main CPU by Scan and Signaling Distributor (SSD) messages. For X3W equipment, the pulses are collected by firmware and complete digits are reported to the main CPU as IPE digit messages.

Operating parameters

X3W trunk cards can only be configured on IPE shelves.

Trunk-to-trunk connections are supported, but the Automatic Number Identification (ANI) information will refer to the ANI DN of the incoming route, except with QSIG, Q931, and Digital Private Signaling System #1 (DPNSS1) routes. QSIG, and Q931 ANI information will use the Calling Line Identification (CLID) information, whereas DPNSS1 ANI will use the Originating Line Identifier (OLI) information if this information is present.

The Dynamic Loss Switching feature is not supported, because there is no connection matrix and loss alternative table available for the CIS market. However, Dynamic Loss Switching is supported in Australia, New Zealand, Italy, and China.

The Static Loss Plan Download (SLPD) feature is supported on X3W trunks.

No loss downloading/switching is done for E3W trunks.

ANI is only supported for outgoing calls.

The data in ANI is built only once at the beginning of the call. Once the trunk access code is dialed, the ANI information is downloaded to the 3WT firmware. The download of ANI occurs only once and is not changed or redownloaded for any kind of operation during a call; therefore, if the call goes through any type of modification such as a transfer or call forward for instance, the ANI information sent when requested is that of the original originator of the call.

Toll Operator Manual Ringing and Break-In are not supported on IPE analog trunks.

Data calls are supported, but with the limitations due to the 500 Hz ANI requests that can happen any time during the call and the ANI information being sent on the same voice circuit on which the data is being transmitted; therefore, the transmission of data is not guaranteed.

Multifrequency Shuttle signaling is not supported on X3W trunk cards.

The CIS A-law XCT (NTD17AE) is required.

Feature interactions

Authorization Code

An extension may, referring to the Authorization Code, seize an outgoing CIS 3WT trunk. The Authorization Code category is used to build the ANI message, meaning that a telephone which has a CIS restricting call category can complete a call to the public network using the Authorization Code.

Autodial

Autodial on a E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange/Central Office, which is expected to appear after the toll access code is dialed. The Public Exchange then does not accept the call due to failure to receive ANI information.

Dial Tone Detection

Dial Tone detectors are supported with the limitations of the reliability of the tone provided by the Public Exchange.

DPNSS1 Gateway

The ANI information transmitted for this incoming DPNSS1 route will include the Local Exchange Code (LEC) of the CIS outgoing route, the ANI DN, and the Category Code (CAC) of this incoming route.

The ANI DN information which is built will refer to the Originating Line Identifier (OLI) if present and the Route DN Length prompt for ANI (RDNL > 0) in LD 16. If the OLI is available, but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the OLI is available, but RDNL = 0 and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the OLI is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the Additional Digit (ADDG). If RDNL > 0, its value will be the number of digits extracted from the OLI to be used as the ANI DN. The least significant digit of the OLI will be extracted (for example, if the DN is 4201, the 1 is the least significant digit.)

If there is no OLI, the ANI DN of the DPNSS1 route is used to build the ANI message. If there is no ANI DN on the DPNSS1 route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDGs of the CIS route (ADDG is always defined).

Incoming Digit Conversion Enhancement, Incoming DID Digit Conversion

The construction of an ANI message does not care if Incoming Digit Conversion is used. The DN sent as ANI is the actual DN of the 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.

Last Number Redial

Last Number Redial on an E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Multiple Appearance Directory Number

Since the ANI category is defined for each telephone for Three Wire Analog Trunks, two stations with the same multiple Appearance DN can be assigned different ANI categories.

Q931 Gateway/BRI Gateway

The ANI information transmitted for this incoming Q931 route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

The ANI DN information which is built will refer to the Calling Line Identification (CLID) if present and the Route DN Length prompt for ANI (RDNL > 0) in LD 16. If the CLID is available but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the ADDG. If RDNL > 0, its value will be the number of digits extracted from the CLID to be used as the ANI DN. The least significant digits of the CLID will be extracted (for example, if the DN is 4201, the 1 is the least significant digit).

If there is no CLID, the ANI DN of the Q931 route is used to build the ANI message. If there is no ANI DN on the Q931 route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDG of the CIS outgoing route (ADDG is always defined).

QSIG Gateway

The ANI information transmitted for this incoming QSIG route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

The ANI DN information which is built will refer to the Calling Line Identification (CLID) if present and the Route DN Length prompt for ANI (RDNL > 0) in LD 16. If the CLID is available but RDNL = 0 for that route, the ANI DN is the ANI DN of that incoming route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, the ANI DN is the ANI DN of the CIS outgoing route. If the CLID is available, but RDNL = 0, and the ANI DN of the incoming route is not defined, and the ANI DN of the CIS outgoing route is not defined, the ANI DN will be built with the ADDG. If RDNL > 0, its value will be the number of digits extracted from the CLID to be used as the ANI DN. The least significant digits of the CLID will be extracted (for example, if the DN is 4201, the 1 is the least significant digit).

If there is no CLID, the ANI DN of the QSIG route is used to build the ANI message. If there is no ANI DN on the QSIG route, the ANI DN of the CIS outgoing route is used to build the ANI message. If there is no ANI DN on the CIS outgoing route, the ANI is built with the ADDG digits of the CIS outgoing route (ADDG is always defined).

The ANI information transmitted for this incoming QSIG route will include the LEC of the CIS outgoing route, the ANI DN, and the CAC of this incoming route.

R2MFC Calling Number Identification

The incoming R2MFC CNI will not be tandemed if the call is outgoing to a CIS trunk. The ANI built will be the LEC of the outgoing CIS route, the ANI DN of this R2MFC incoming route if defined (otherwise it will be the ANI DN of the outgoing CIS route, or the ADDG digit), and the CAC of this incoming R2MFC route.

The category (CAC) used to build the R2MFC Calling Number Identification (CNI) for the analog, digital and Basic Rate Interface (BRI) sets is used to build the CIS ANI. The meaning of CAC is different between the R2MFC CNI signaling and the CIS signaling (analog BRI, and digital). R2MFC CAC prompt values are in the range of 0 to 10, and the default is 0. CIS CAC prompt values are in the range of 0 to 9, and the default value is 3.

If the MFC package is equipped, but not the CIST package, the CAC prompt uses the R2MFC range and default. If the CIST package is equipped (MFC package equipped or not) the CAC prompt uses the CIS range and default.

Speed Call

Speed Call on an E3W trunk will fail for toll calls. E3W trunks do not wait for the ANI request from the Public Exchange, that is expected to appear after the toll access code is dialed. The Public Exchange will not accept the call due to the failure to receive ANI information.

Virtual Network Services

Virtual Network Services is not supported on CIS trunks.

Feature packaging

The Three-Wire Analog Trunk – CIS feature is contained in Commonwealth of Independent States Trunk Interface (CIST) package 221.

The following packages are also required to implement this feature:

- Fast Tone and Digit Switch (FTDS) package 87 (only for E3W cards)
- Flexible Tones and Cadences (FTC) package 125
- International Supplementary Features (SUPP) package 131 for DID/DOD
- Flexible Numbering Plan (FNP) package 160

- Trunk Failure Monitor (TFM) package 182, and
- Meridian 1 Intelligent Peripheral Equipment (XPE) package 203 (only for X3W cards).

Feature implementation

Task summary list

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

- Table 83: LD 17 on page 278

  Configure the system data.
- Table 84: LD 16 on page 279

  Configure an incoming X3W DID route.
- Table 85: LD 16 on page 280

  Configure an outgoing X3W DID route and define the toll digit using the TDG prompt.
- Table 86: LD 18 on page 281

  Configure the Special Service List.
- Table 87: LD 16 on page 281

  Configure an outgoing X3W DID route and define the toll access code using the SSL prompt.
- Table 88: LD 16 on page 282

  Configure an incoming E3W DID route.
- Table 89: LD 16 on page 283

  Configure an outgoing E3W COT route.
- Table 90: LD 14 on page 284

  Add or change trunk data for X3W incoming DID trunk.
- Table 91: LD 14 on page 285

  Add or change trunk data for X3W outgoing DID trunk.
- Table 92: LD 14 on page 285

  Add or change trunk data for E3W incoming three-wire trunk.
- Table 93: LD 14 on page 286

Add or change trunk data for E3W outgoing three-wire trunk.

- Table 94: LD 10 on page 286

  Add or change analog (500/2500-type) telephones for CIS.

- Table 95: LD 11 on page 287

  Add or change Meridian 1 proprietary telephones for CIS.

- Table 96: LD 12 on page 287

  Add or change an attendant console for CIS.

- Table 97: LD 27 on page 287

  Add or change Basic Rate Interface (BRI) sets for CIS.

- Table 98: LD 56 on page 287

  Configure dial tone, busy tone, and tone to last party.

- Table 99: LD 88 on page 289

  Configure the Authcode data block.

- Table 100: LD 97 on page 289

  Configure the IPE system record for three-wire trunks.

This is an example that describes how the 3WT related features are configured. Only the prompts that are significant for the Three-Wire Analog Trunk – CIS feature are mentioned.

The following features are needed to make the feature work according to this example: B34 Codec Static Loss Plan Downloading; Partial Dial Timer; End-of-Selection Busy; Tone to-Last Party; Special Dial Tones After Dialed Numbers; Trunk Barring, and Special Service List.

Table 83: LD 17

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	PARM	Gate opener.
- PCML	A	System Pulse Code Modulation companding law. A-law is to be used in the CIS market.
...		
- DTRB	70	Dual-tone Multifrequency burst and interdigit pause for the Tone and Digit Switch. Pulse/Pause Ratio 70/70. For outgoing E3W cards, the preferable digitone burst time is 70 ms.

Table 84: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	ICT	Incoming trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0. Validation is performed by 3WT firmware.
- TIMR	GTI 128	Incoming guard timer.
- TIMR	EOD 13952	End of dial timer, default value in milliseconds.
- TIMR	DSI 11904	Disconnect supervision timer in milliseconds.
- TIMR	DDL 0	Delay Dial Timer not needed.
...		
NEDC	ORG	Near End Disconnect Control. Originating end control.
FEDC	ORG	Far End Disconnect Control. Originating end control.
CDPC	(NO)	The system is not the controlling party on incoming calls.
...		
OPR	(NO)	This is not an outpulsing route.
PRDL	YES	Partial dial timing is equipped using EOD.
EOS	BSY	Busy signal is sent on time-out.
DNSZ	(0)-7	Number of digits expected on DID routes. 0, the default, indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Busy Tone Time. Length of Busy/overflow to be returned on DID routes in seconds.
...		
CAC	0-(3)-9	Route ANI category.

Prompt	Response	Description
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

Table 85: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ATO 128-(4992)-65408	ANI time out timer in milliseconds. For CIS outgoing trunk routes this defines the time delay performed after the outpulsing of the toll access code.
- TIMR	OGF 0	Outgoing flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
- TIMR	GTO 2944	Outgoing guard timer.
...		
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
...		
NATL	NO	North American Toll scheme.
TDG	8	Toll Digits. List of digits after trunk access code which indicate toll calls.
...		

Prompt	Response	Description
OPR	(NO)	This is not an outpulsing route.
...		
ACKW	(NO)	Seizure acknowledge signal is not expected.
...		
LEC	0-9999999	Local Exchange Code. A value must be entered.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

Table 86: LD 18

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	SSL	Special Service List data block.
CUST	xx	Customer number, as defined in LD 15
SSL	1-15	List number for Special Service List.
SSDG	xxxx	Special Service Digit or Digits (1 to 4 digits).
...		
- TOLL	YES	The SSDG entry is a toll number.
...		
SSDG	xxxx	Special Service Digit or Digits (1 to 4 digits).
...		
- SSUC	YES	The SSDG entry is a Special Service unanswered call.
SSDG	<CR>	

Table 87: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.

Prompt	Response	Description
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
...		
SSL	1	Special Service List number.
...		
LEC	0-9999999	Local Exchange Code.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

Table 88: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	DID	Direct Inward Dialing trunk data block.
...		
DTRK	NO	This is not a digital trunk route.
...		
ICOG	ICT	Incoming trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0. Validation has already been done by 3WT firmware.

Prompt	Response	Description
- TIMR	OGF 0	Outgoing flash timer should be set to 0. Validation has already been done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
...		
NEDC	ORG	Near End Disconnect Control Originating end control.
FEDC	ORG	Far End Disconnect Control Originating end control.
CDPC	(NO)	The system is not the controlling party on incoming calls.
...		
OPR	(NO)	This is not an outpulsing route.
PRDL	YES	Partial dial timing is equipped using EOD.
EOS	BSY	End of selection and busy signals enabled.
DNSZ	(0)-7	Number of digits expected on DID routes. 0, the default, indicates no fixed value. This value must be defined according to the numbering plan.
...		
BTT	30	Length of busy/overflow tone to be returned on DID routes in seconds.
...		
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

Table 89: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
...		
TKTP	COT	Central Office Trunk data block.
...		
DTRK	NO	This is not a digital trunk route.

Prompt	Response	Description
...		
ICOG	OGT	Outgoing trunk.
...		
CNTL	YES	Change control or timers.
- TIMR	ICF 0	Incoming flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	OGF 0	Outgoing flash timer should be set to 0 in milliseconds. Validation will be done by 3WT firmware.
- TIMR	EOD 13952	End of dial timer, default value.
- TIMR	DSI 11904	Disconnect supervision timer.
- TIMR	DDL 0	Delay Dial Timer not needed.
- TIMR	GTO 2944	Outgoing guard timer.
...		
NEDC	ETH	Near End Disconnect Control Either end control.
FEDC	ETH	Far End Disconnect Control Either end control.
CDPC	(NO)	The system is not the controlling party on incoming calls.
...		
NATL	NO	North American Toll scheme.
...		
LEC	0-9999999	Local Exchange Code.
ADDG	0-(8)-9	Additional digit.
CAC	0-(3)-9	Route ANI category.
ANDN	0-9999999	Route ANI DN.
RDNL	0-(4)-7	Route DN Length for ANI. This is printed for DPNSS1, MCDN, and QSIG routes only.

Table 90: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
XTRK	XDID	Extended Trunk Type. IPE DID trunk card.
...		

Prompt	Response	Description
SIGL	CIS	Trunk Signaling. Three-wire CIS trunk signaling.
CIST	(NO) YES	Prompted only for incoming routes (that is, ICOG = ICT). NO = Local trunk. YES = Toll trunk.
...		
STRI	IMM	Immediate incoming start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP) (SHL) LOL (BARD) BARA	Dial pulse (for 3WT incoming and outgoing). Line length used for pad settings. Barring (denied) allowed.

Table 91: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
XTRK	XDID	IPE DID trunk card.
...		
SIGL	CIS	Three-wire CIS trunk signaling.
...		
STRO	IMM	Immediate outgoing start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP) (SHL) LOL (BARA) BARD	Dial pulse (for 3WT incoming and outgoing). Line length used for pad settings. Barring (allowed) denied.

Table 92: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DID	Direct Inward Dialing trunk data block.
...		
SIGL	EAM	Ear & mouth.

Prompt	Response	Description
CDEN	DD	Double density.
...		
STRI	IMM	Immediate incoming start arrangement.
...		
SUPN	YES	Answer and disconnect supervision required.
CLS	(DIP)	Dial pulse.

Table 93: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	COT	Central Office Trunk data block.
...		
SIGL	LOP	Loop start.
CDEN	DD	Double density.
...		
SUPN	YES	Answer and disconnect supervision required.
- STYP	PSP	Polarity sensitive card.
...		
SEIZ	YES	Answer and disconnect supervision required.
CLS	DTN	Digitone.

Table 94: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	analog (500/2500-type) telephone data block.
...		
CLS	(DNAA) DNAD	DN of telephone (allowed) denied for use in ANI messages.
CAC_CIS	0-9	Specifies ANI category for 3WT calls.

Table 95: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS	(DNAA) DNAD	DN of telephone (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

Table 96: LD 12

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	2250	Attendant console type.
...		
CLS	(DNAA) DNAD	DN of telephone (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

Table 97: LD 27

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	DSL	Digital Subscriber Loop data block.
...		
CLS	(DNAA) DNAD	DN of telephone (allowed) denied for use in ANI messages.
CAC	0-9	Specifies ANI category for 3WT calls.

Table 98: LD 56

Prompt	Response	Description
REQ	NEW CHG PRT	Add, change, or print.
TYPE	MCAD	Master Cadence data block.
WACD	30	Cadence number. In this example entry 30 is modified.
CDNC	60 60	On-off phases for cadence.

Prompt	Response	Description
REQ	NEW CHG PRT	Add, change, or print.
TYPE	FCAD	Firmware Cadence data block.
WACD	30	Cadence number. In this example entry 30 is modified.
CDNC	60 60	On-off phases for cadence. 0.3 second on, 0.3 second off.
END	REPT	Repeating cycles.
- CYCS	1	On/off cycles to be repeated.
- WTON	YES	Define tones associated with the cadence.
- - TONES	158	420 Hz and -12 dB below overload.
REQ	NEW CHG PRT	Add, change, or print.
TYPE	FTC	Flexible Tones and Cadences data block. Used to provide special dial tone after dialed number.
...		
HCCT	YES	Hardware Controlled Cadences and Tones modification of the hardware.
...		
- BUSY		Busy tone.
- - TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	30	XCT cadence number. 0.3 seconds on, 0.3 seconds off.
...		
- TLP		Tone to last party.
-- TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	30	XCT cadence number. 0.3 seconds on, 0.3 seconds off.
- TLTP	30	Tone to last party timer in seconds.
...		
SRC	YES	Source Tones.
- SRC1		CIS continuous dial tone within the range.
- - TDSH		
- - XTON	158	420 Hz and -12 dB below overload.
- - XCAD	0	No cadence.

Prompt	Response	Description
REQ	NEW CHG PRT	Add, change, or print.
TYPE	DTAD	Special Dial Tone After Dialed Number data block.
DDGT	9	The digit 9 is to be used as an outgoing local access code.
TONE	SRC1	Tone to be provided after the dialed digit 9.

Table 99: LD 88

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	AUB	Authcode data block.
...		
CLAS	(0)-115	Classcode value assigned to Authcode (NAUT).
...		
NCOS	(0)-99	Network Class of Service Group number.
CAC_CIS	0-9	Specifies ANI category for CIS calls.

Table 100: LD 97

Prompt	Response	Description
REQ	CHG	Change.
TYPE	LOSP	Loss Plan Tables. Configure loss parameters for downloading.
...		
TTYP	(STAT)	Install a B34 Static Loss Plan Table.
- STYP	(PRED)	A numbered predefined table is to be used.
- - TNUM	28	28 = CIS Table.
REQ	CHG	Change.
TYPE	LOSP	Loss Plan Tables. Configure loss parameters for downloading.
...		
TTYP	(STAT)	Install a B34 Static Loss Plan Table.
- STYP	CSTM	Customize a numbered predefined table.
PWD2	xxxx	Response CSTM at STYP prompt requires a PWD2 password or a LAPW password with Loss Planning

Prompt	Response	Description
		Customizing Allowed (LOSA) access. This prompt appears if the appropriate password has not been given previously.
- DIDS	Rx Tx	Enter loss levels for DID short line.
- DIDL	Rx Tx	Enter loss levels for DID long line.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 41: Time and Date

Contents

This section contains information on the following topics:

Feature description

The Time and Date feature provides the capability to display or modify the system time and date from the attendant console. If Display Time or Display Date keys are installed on the console, pressing the respective key causes the time or date to be shown on the digit display. However, these keys only allow information to be displayed, not changed.

The Change Time or Change Date keys allow the attendant to change the time or date. When a change is made, the system clock is altered to the new values. The change keys also allow display of the time or date.

Operating parameters

The Time and Date feature is available with M2250 consoles.

If the Change Time (MTM) and Change Date (MDT) keys are provided on a console, there is no need for the Display Time (DTM) and Display Date (DDT) keys because the MTM and MDT keys provide the display capability. DTM and DDT keys are used when the console is only allowed to view, but not change, the time and date.

When using the MTM and MDT keys, the date must be entered in the day, month, and year format; and the time must be entered in the 24-hour clock format. This is true even if the M2250 has selected a different date and time format.

The M2250 console continuously shows the time and date on line one of the display. The attendant can change the format of time and date by using the Options menu. The date and time are downloaded to the M2250 console from the system clock and cannot be changed by the Options menu. The change time and date keys are required.

A call cannot be answered while the display/change key is activated; however, the keys can be used once the call is established.

Feature interactions

Hold

Loops used when updating time or date cannot be put on hold.

In-Band Automatic Number Identification

If the agent presses the Time and Date (TAD) key while on an In-Band Automatic Number Identification (IANI) call, the time and date remain displayed throughout the call. To display the ANI number again, place the call on hold and retrieve it. The ANI number reappears.

Network Time Synchronization

As done with the LD 12, every time the Time and Date Attendant key is used to change the system time, a request for synchronization will be made to the Master to accurately set the seconds.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 101: LD 12 - Assign Time and Date keys on attendant consoles.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx DDT xx DTM xx MDT xx MTM	Add a Display Date key. Add a Display Time key. Add a Display/Change Date key. Add a Display/Change Time key. The range of key numbers (xx) is 0-19 on the M2250 console, and 0-9 on all other consoles.

Feature operation

To view the Time, press **Display Time (DTM)** .

To view the Date, press **Display Date (DDT)** .

To change the time, follow these steps:

1. Select an idle loop key.
2. Press **Change Time (MTM)** .
3. Enter the time using the 24-hour clock for hours and minutes (00 00).
4. Press **Change Time (MTM)**.
5. Press **Rls**.

To change the date, follow these steps:

1. Select an idle loop key.
2. Press **Change Date (MDT)**.
3. Enter the date using two digits for day, month, and year (dd mm yy).
4. Press **Change Date**.
5. Press **Rls**.

Chapter 42: Tone to Last Party

Contents

This section contains information on the following topics:

Feature description

This feature allows a Tone to Last Party (TLP) tone to be sent to analog (500/2500-type) telephones or trunks that are in the half disconnect state. The TLP is given until the system releases the trunk, or the TLP timer (0-32 seconds) times out.

During the time that the TLP tone is given to the telephone, the telephone appears busy to all incoming calls. Camp-on is denied, and attendant Break-in, busy verify, and override are temporarily denied during this time.

If a telephone is not placed on-hook and the timer times out, the telephone is in line lockout state, and remains so until it is placed on-hook.

A trunk is in the half disconnect state if the near-end has disconnected, but the system is still holding the trunk, waiting for a message from the far-end, or for the disconnect supervision timer to time out. Barge-in is denied while the trunk is receiving the TLP tone.

The TLP is defined in each tone table. The TLP for analog (500/2500-type) telephones is defined on a customer basis, while the TLP for trunks is defined on a route basis.

Operating parameters

The TLP tone is not given to a telephone that is receiving another tone.

This feature does not apply to service trunks, such as music, paging and recorded announcement.

The TLP tone is not given to a trunk if it is being held because of the guard timer.

Feature interactions

Multi-Party Operations

The TLP tone is not given to a telephone which has Multi-Party Operations (MPO).

Operator Call Back China 1

Operator Call Back China #1 (OPCB) has precedence over TLP.

Feature packaging

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

Feature implementation

Table 102: LD 56 - Modify or change customer tone and ringing parameters.

Prompt	Response	Description
...		
TLP	ccc ttt x xx xx xx	Tone to Last Party.

Prompt	Response	Description
TLPT	(0)-32	Tone to Last Party Timer (seconds). No tone is given if TLPT = 0.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 43: Tones and Cadences

Contents

This section contains information on the following topics:

Feature description

A tone is the frequency and level of the sound produced while the telephone is ringing, providing dial tone, or providing feature activation tones. A cadence defines the time duration for the on and off phases of a ringing or tone cycle.

A collection of basic tones and cadences is available on all systems. Flexible Tones and Cadences (FTC) package 125 allows the tones to be changed.

Basic Tones and Cadences

Special dial tone

Special dial tone is supplied by the system to indicate a request for Call Transfer, Conference, and Ring Again. Special dial tone differs from regular dial tone in that it has three 128 ms interruptions at the beginning of the tone.

Overflow tone

Overflow tone can be provided on an optional basis to a station user who tries to access a trunk group when all trunks are busy, or who attempts to access features that are unavailable to their telephone. Overflow tone is best described as a fast busy signal.

Tone buzzing

Tone buzzing is used in conjunction with such features as Call Waiting and Manual Signaling (Buzz) to alert the user by a buzz tone through the telephone's loudspeaker. This applies when the telephone is off-hook or has a headset plugged in.

Flexible Tones and Cadences

The Flexible Tones and Cadences (FTC) feature, allowing the system to adapt to the tone specifications of different countries. Tones such as dial, special dial, busy, ringback, overflow, test, normal, and distinctive ringing are hardware controlled from the Tone and Digit Switch (TDS) circuit card (see Table 103: Hardware controlled tones on page 300). Tones such as camp-on, call waiting, intrusion, and override are software controlled, although the basic tone is still coming from the TDS card (see Table 104: Software controlled tones on page 301).

The desired cadences for the software controlled tones are defined by providing the system with the time length of the on and off phases. Software also controls ringing for analog (500/2500-type) telephones, although the voltage is supplied by the ring generator card.

The tone data is stored in tables. Every customer and route must select which tone table to use. Table 0 is filled in with default hexadecimal codes when the first customer is created and must not be changed.

All data related to the flexible tones is kept in isolated areas called Flexible Tone tables. Software Cadence tones and Master Cadence tables have an index into the MCAD table for its corresponding software cadence.

Most of the cadences are expressed in multiples of five milliseconds (ms). Therefore, in addition to the existing 128 ms timing mark, a 96 ms timing mark is introduced by a new read only memory (ROM) pack with new firmware.

Table 103: Hardware controlled tones

Tone	Description
Dial tone	Indicates the system can accept dialing.
Message Waiting dial tone	Indicates a message is waiting at the message center.

Tone	Description
Call Forward dial tone	Indicates that the user has call forwarded the telephone.
Call Forward Message Waiting dial tone	Indicates that the user has call forwarded the telephone and a message is waiting at the message center.
Control Dial tone	Used for broker service to indicate a control digit is required after the switchhook (only for 2500-type telephones with Digitone class of service).
Busy tone	Indicates that the called DN is busy.
Ringback tone	Given to the calling party while the called party is ringing. Also given to Central Office trunks waiting for the DN to answer.
ACD RGA Ringback tone	Given to a caller to an Automatic Call Distribution (ACD) group when entering the waiting call queue and having RGA (Ring Again).
Overflow tone	Indicates that the trunk route is busy, or the DN is blocked or disabled, or that a not-allowed action has been carried out.
LDN tone	Indicates to a Centralized Attendant Service (CAS) attendant that the incoming call is a Listed DN (LDN) call from a remote site.
Camp-On tone	Provided as an initial burst when the attendant extends a call to a busy DN that is not equipped with the Call Waiting feature.
Camp-On Confirm tone	Confirms to a CAS attendant that a call to a busy DN at remote site has camped on, or that the called DN has not answered after a specified time and the calling party has come back.
Dial "0" Recall tone	Indicates to a CAS attendant that a call is a recall occurring due to attendant recall or call forward busy to an attendant from a remote site.
Hold Confirm tone	Indicates to a CAS attendant that a call placed on silent hold has timed out and is recalling.
Test tone	Provided during testing of trunk circuits.
Distinctive Ring tone	Used to differentiate between routes.
Normal Ring tone	Provided for internal calls and incoming calls if distinctive ringing or precedence ringing is not in use.

Table 104: Software controlled tones

Tone	Description
Agent Observe tone	Given to an agent being observed by a supervisor.
Call Waiting tone	Indicates to a busy station that another call is coming in.
Intrusion tone	Provided when the attendant initiates the Barge-In, Busy Verify, or Break-In feature.

Tone	Description
Override tone	Provided when a user operates the Override key and enters the conversation of a busy extension.
Observe Blocking tone	Given to the supervisor who encounters blocking while attempting to observe an agent.
Off-Hook Queuing tone	Given to the call originator when the call enters the off-hook queue.
Set Relocate tone	Given after all information needed to relocate the telephone is given and proven to be correct. Also given to indicate all is correct after plugging the telephone back in at the relocated Terminal Number (TN).
Telset Messaging Alert tone	Indicates to caller that Telset messaging facilities have been entered.
Telset Messaging OK tone	Indicates to caller that the message has been received correctly and everything is fine.
Tel Status Update tone	Indicates a successful status update process.
Special Dial tone	Indicates the availability of a special function such as Conference or Transfer.
Expensive Route Warning tone	When Automatic Route Selection is in use, indicates that all inexpensive routes are busy and an expensive route must be chosen to complete the call.
ACD Call Force tone	Indicates to the ACD agent that the current call has been disconnected and a new caller is about to be given to the agent.

Operating parameters

The tones that can be produced are limited to the tones available on the particular TDS card being used.

Gradual level change is not allowed when a tone is activated.

If the Distinctive Ringing package is equipped, and a trunk route is classmarked for that feature, the cadence chosen for each call comes from the same tone table as for a normal call. The Distinctive Ringing field determines the cadences.

If a parked call was originally distinctive, and FTC is equipped, the Call Park Recall cadence takes precedence. If FTC is not equipped, the distinctive precedence ringing is given.

Because Enhanced Flexible Tones and Cadences (EFTC) is an enhancement to Flexible Tones and Cadences (FTC), the FTC package must be equipped.

A customer option determines whether the cadence will be defined by the originating or the terminating end of the call.

Feature interactions

Audible Reminder of Held Call

This feature allows for a definable cadence as a reminder of a held call. With an analog (500/2500-type) telephone, the cadence is determined by the customer's Flexible Tones and Cadence (FTC) table for the holding party. Ringing on an analog (500/2500-type) telephone is not affected by definitions for the Incoming Route option. The cadence for the reminder, and the duration between reminder rings, is always defined within the customer's tone table.

Call Forward Reminder Tone

The Call Forward Reminder Tone feature provides a way to determine whether the call forwarding feature on an analog (500/2500-type) telephone is active. For systems equipped with the FTC package, the Call Forward Reminder Tone Allowed option gives the dial tone defined by Call Forward Dial Tone to an analog (500/2500-type) telephone that has Call Forward active with no message waiting and the dial tone defined by Call Forward Message Waiting to an analog (500/2500-type) telephone that has Call Forward active and a message waiting. To get different Call Forward and Call Forward Message Waiting reminder dial tones, it is necessary to define a distinct tone and cadence for Call Forward Dial Tone and a distinct tone and cadence for Call Forward Message Waiting in LD 56, as well as to specify Call Forward Reminder Tone Allowed in LD 15.

Call Park Recall and Group Call Ring

Recall Ring and Group Call Ring are given special entries in the FTC table. New entries are added to the FTC overlay (LD 56) to define the cadence for Meridian 1 proprietary telephones, and analog (500/2500-type) telephones. The new Recall Ring entry is used to ring a telephone when recalling a Parked Call.

Conference Warning Tone Enhancement

There are no changes to the limitations to cadence numbers entry values. The same restriction still applies.

Ringing Based on Incoming Route

Enhanced Flexible Tones and Cadences (EFTC) allows the route's tone table to determine the cadence and ringing frequency for incoming calls.

10-Phase Cadence

Programming of software controlled cadences expands with EFTC from 4 intervals to 10, offering greater versatility with the cadences and cadence phases. This affects all cadences under software control.

Feature packaging

Flexible Tones and Cadences (FTC) package 125 has no feature package dependencies.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 44: Tones, Flexible Incoming

Contents

This section contains information on the following topics:

Feature description

When a telephone is off-hook, the user is alerted to a second incoming call by a buzz tone. Flexible Incoming Tones (FIT) allows the replacement of the standard buzz tone with a buzz with an on/off cadence. This feature is defined on an individual telephone basis.

When a call is presented to a telephone in any of the following situations, a tone with a special cadence alerts the user:

- Call on DN key while busy on another DN
- Call to a station that is off-hook
- Call Park recall when station is busy on another DN
- Call on Group Call key while busy on another call
- Call Waiting, and
- Call on Dial Intercom key while busy on another call.

The buzz cadence is the same as the ringing cadence that applies to a particular kind of call. For example, if a user receives a call that is a Group Call, FIT alerts users with a buzz cadence unique to group calls. If the user receives a call on the Call Waiting key, FIT provides a buzz cadence signifying call waiting.

Operating parameters

Flexible Incoming Tones applies only to Meridian 1 proprietary telephones.

Flexible Incoming Tones does not apply to the following:

- Automatic Call Distribution (ACD) call forcing
- ACD agent receiving a call on ASP key
- ACD supervisor receiving a call on AMG key
- Manual signaling
- Signal Source activated by an attendant console, and
- Ring Again.

Digital telephones in Handsfree mode receive the regular buzz, even if FIT is enabled.

The telephone buzzes with a cadence only if the customer and telephone options are activated. If either option is off, the telephone receives the standard buzz.

Feature interactions

Automatic Call Distribution

If an Automatic Call Distribution (ACD) agent telephone has FIT allowed and either is off-hook in the handset mode or has the headset plugged in, the agent receives a buzz cadence when a new call is presented. If FIT is not allowed, the agent telephone receives the standard buzz tone.

Dial Intercom Groups

For Dial Intercom Group (DIG) calls with the voice (V) option, if the telephone receiving the call is busy, the user hears one buzz followed by a flashing indicator. This is how DIG works with or without FIT.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 105: LD 15 on page 307

   Allow or deny Flexible Incoming Tones (FIT) at the customer level.

2. Table 106: LD 11 on page 307

   Allow or deny Flexible Incoming Tones for Meridian 1 proprietary telephones.

Table 105: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- OPT	(DBD) DBA	FIT (denied) allowed for Meridian digital telephones.

Table 106: 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	(FITD) FITA	Flexible Incoming Tone (denied) allowed.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 45: Total Redirection Count

Contents

This section contains information on the following topics:

Feature description

This feature allows a limit to be defined on the number of redirections within a private network allowed to a call, before the call receives intercept treatment.

Both the limit on the redirection for a call and the type of Intercept treatment that the call receives are customer-defined in LD 15. This applies to on-node and off-node redirections, and to all types of redirections.

Operating parameters

The maximum value that may be given to the Total Redirection Count (TRCNT) limit is seven.

The TRCNT is kept active until the call is established or directed to the attendant.

The TRCNT takes precedence over higher count limits placed on redirected calls, while lower count limits take precedence over the TRCNT.

It is possible to define a different TRCNT limit at each node. For this reason, it is possible for a node to receive a redirected call from another node that exceeds its TRCNT limit. In this case, the TRCNT count for the call is set to the TRCNT limit defined for the node. At least one attempt is made to terminate the call before intercept treatment is given.

For off-node operation, the TRCNT count overrides the Redirection Count (RCNT) count in the Integrated Services Digital Network (ISDN) field in the SETUP message. This implies that the count transmitted to a node is either interpreted as TRCNT or Call Redirection Threshold (RCNT), depending on the configuration at the receiving node.

For off-node calls, this feature applies only to systems using Meridian Customer Defined Networking (MCDN) signaling over ISDN Signaling Link (ISL)/ISDN TIE links. Network Attendant Service is required to route a call to an attendant at another node.

Intercept to the attendant does not count as a redirection attempt.

The following ISDN call restrictions apply:

- Tandem Threshold, which is the limit placed on the number of tandem nodes allowed in a network connection
- The Public Service Telephone Network (PSTN) Threshold, which is the limit placed on the number of PSTNs allowed in a network connection
- The Call Redirection Threshold, which is a limit on the number of times that a call can be redirected off-node. If the Total Redirection Count (TRCNT) Limit is a value greater than zero, the ISDN field in the SETUP message transports the TRCNT information rather than the Redirection Count (RCNT) information
- The Mµ/A Law Conversion Threshold, which is a limit on the number of Mµ/A Law Conversions allowed in a network connection
- Satellite Delay Threshold, which is a limit on the number of satellite delays allowed in a network connection
- Disconnect Supervision Threshold, which limits to one the number of unsupervised trunks allowed in a network connection

Feature interactions

Call Forward No Answer and Transfer

If a call has attempted Call Forward No Answer and was extended by the attendant, the call will not be intercepted when the TRCNT limit has been exceeded. The call will continue to ring the telephone until recalled to the attendant.

If Overflow (OVF), Busy (BSY), or Source (SRCx) is configured as Intercept Treatments a call attempting Call Forward No Answer, that exceeds the Total Redirection Count limit, will not be intercepted. Further redirections are prohibited and the call continues to ring the current telephone.

Group Hunt

Group Hunt takes precedence over the TRCNT feature, in that the TRCNT limit is not applied to a Group Hunt call.

Hunt, Call Forward Busy, Call Forward All, Calls Call Forward No Answer, Second-level Call Forward No Answer

Hunt, Call Forward Busy, Call Forward All Calls, Call Forward No Answer, and Second-level Call Forward No Answer redirections are limited to the value defined in the TRCNT limit (if greater than 0). If this limit is exceeded, intercept treatment is given.

Intercept treatment

Intercept treatment is not given if a call is a Network Automatic Call Distribution (NACD) ACD call, if a call is a Central Office trunk in Night Service (specific treatment is given rather than customer-defined intercept treatment), or if the call is a data call (overflow tone is automatically given).

Feature packaging

For internode operation, Integrated Services Digital Network (ISDN) package 145.

For detecting trunk type across a network, Network Attendant Service (NAS) package 159.

For attendant display, Calling Party Name Display (CPND) package 95.

For the attendant to override a redirection configuration, Attendant Break-in/Trunk Offer (BKI) package 127.

Feature implementation

Use Table 107: LD 15 on page 312to configure the type of intercept treatment that the redirected call receives, and the number of times that a call can be redirected before being intercepted.

Table 107: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	INT	Intercept treatment options
...		
- RCLE	(ATN) OVF ATN	Redirection Count Limited Exceeded as defined by TRCL. ATN is not allowed for attendant calls. NAP is not allowed for any field for RCLE.
TYPE	RDR	Call Redirection
...		
- TRCL	(0)-7	Total Redirection Count Limit. Number of times that a call can be redirected before being intercepted. Zero means that redirection is not limited by this feature, but is limited by various configurations.

Feature operation

When the total redirection count exceeds the defined limit, the call receives the customer-defined intercept treatment. This treatment includes receiving busy indication, overflow indication, or recorded announcement, receiving one of eight special tones, or being routed to the attendant. If the call is routed to the attendant, it is presented on the Incoming Call Indicator (ICI) Intercept key and the reason for redirection is given on the console display. The attendant may then use Attendant Break-In to connect to the desired station (if the desired station is established on a call).

Chapter 46: Trunk Barring

Contents

This section contains information on the following topics:

Feature description

The Trunk Barring feature provides the option of denying or allowing a direct or modified connection between customer defined routes.

Trunk Barring works in conjunction with Route Access Restriction Tables (ARTs) defined in LD 56. Trunk Barring is applied on a route basis. The four route categories that Trunk Barring recognizes, and the types of routes in each category, appear in the following table:

Table 108: Route categories and types recognized by Trunk Barring

Route Category	Route Types
Central Trunk Office (COT)	COT, FEX, WAT
Direct Inward Dialing	DID
TIE	TIE, CAA, CAM, CSA
Other trunk types	ADM, DIC, MDM, PAG, RCD

Trunk Barring applies to all methods of connecting the trunks (for example, dialing route access, call modification, or attendant extension). A route is allocated an Access Restriction Table (ART) linked by a table number (ART number) in the Route Data Block. The ART to be used for a connection is determined by the first trunk in the connection independent of whether the trunks are incoming or outgoing. The first trunk in the connection is referred to as the Originating Trunk Connection (OTC).

A default table exists so that LD 56 does not have to be used to assign an ART number to a newly created route. If the default value for each Route Category is ART number 0, no trunk barring will occur.

Operating parameters

When activated in conjunction with the Route Access Restriction Tables, Trunk Barring prohibits previously allowed connections. Previously restricted connections cannot be lifted or circumvented by Trunk Barring.

Trunk Barring does not apply to Recorded Announcement (RAN), Music (MUS), Automatic Wake-Up (AWU), or Centralized Attendant Service (CAS) trunks as it is inconsistent with their defined purposes.

Feature interactions

Access Restrictions

Trunk Barring is at the top of the hierarchy for access restrictions.

Attendant Break-In

Trunk Barring does not result in intercept treatment for Toll Operator Break-In.

Attendant-Extended Calls

When an attendant attempts to extend an Originating Trunk Connection on a barred route, overflow tone is given.

Call Forward

This section applies to Call Forward All Calls, Call Forward Busy, Call Forward by Call Type, Call Forward External Deny, Call Forward Internal Calls, Call Forward No Answer, Call Forward No Answer Second Level, and Call Forwarding

If an Originating Trunk Connection is forwarded to a barred route, the caller receives the intercept treatment specified in the Customer Data Block.

Call Transfer

The originator of a call transfer, unless otherwise restricted, is able to connect to a denied party on a consultation basis. Operating the Transfer key on a Meridian 1 proprietary telephone or going on-hook on an analog (500/2500-type) telephone does not result in a call transfer if the Originating Trunk Connection is barred. The user of a Meridian 1 proprietary telephone remains connected to the denied party until releasing the connection and returning to the held Originating Trunk Connection. The user of an analog (500/2500-type) telephone is re-rung by the Originating Trunk Connection when a transfer is attempted and denied.

Conference Calls

The originator of a conference call can only connect to a barred route on a consultation basis. A switchhook flash from an analog (500/2500-type) telephone results in a reestablished connection with the Originating Trunk Connection. The user of a Meridian 1 proprietary telephone must release the barred connection to return to the Originating Trunk connection, or the conference containing the Originating Trunk connection; operating the Conference key on a Meridian 1 proprietary telephone has no effect. An attendant can return to the Originating Trunk Connection, or the conference containing the Originating Trunk Connection, by releasing the barred connection. This is done by pressing the RLS DEST key; pressing the Conference key has no effect.

Direct Trunk Access

When an Originating Trunk Connection attempts a trunk connection to a route which is restricted by its Access Restriction Table, the connection is not allowed. The intercept treatment specified in the Customer Data Block is applied.

Enhanced Night Service

Any incoming call that is routed by Enhanced Night Service to a telephone from which it is barred will not be connected. Overflow tone (fast busy) will be given to the incoming trunk instead.

Intercept Treatment

A telephone that is intercepted to the attendant cannot apply Ring Again on No Answer.

ISDN Semi Permanent for Australia

For calls using or requesting an ISPC link, Trunk Barring is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS)

If one route is barred, the system will look for the next route in the Route List Index (RLI) and if this route is not barred, the call will go through on this route. If the second route is barred, the system will continue searching the next route in the Route List Index, until an unbarred route is found.

When implementing Trunk Barring caution must be exercised not to circumvent the intended NARS/BARS restrictions.

Toll Operator Break-In

Trunk Barring results in intercept treatment for all route types that can be barred, except Toll Operator Break-In.

Trunk to Trunk Connection

Trunk Barring takes precedence over the Trunk to Trunk Connection feature.

Virtual Network Services

With respect to this feature the following cases apply:

- When the second trunk involved in the call is used by VNS, no trunk barring is applied regardless of the configuration of the first trunk. The call is always allowed to get through.

  This implementation completely overrides the Trunk Barring feature.
- When the first trunk involved in the call uses VNS, and the second one is not used by VNS, trunk barring is performed according to the content of the default ART table for the TIE trunk.

Feature packaging

This feature requires Trunk Barring (TBAR) package 132.

Feature implementation

Task summary list

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

1. Table 109: LD 56 on page 319

   Modify trunk barring Access Restriction Tables (ARTs).
2. Table 110: LD 56 on page 320

   Change or print ART number for the route.
3. Table 111: LD 56 on page 320

   Change or print the route category default table.

In most cases that require barring, only one ART is necessary, although multiple ARTs may be defined per route. Whenever a new route is created (in LD 16), the default ART defined for that route type is assigned to the route. This default depends on the route type being created.

The flexibility of assigning ART by route is also available. The default table which specifies which ART table is to be assigned to a route type is changeable in LD 56. Until this is done, the default ART is used.

The following is a guideline on how to configure Trunk Barring:

1. Gather all information regarding the type of route to be used in the system.
2. For each route type, list beside it the route types that are barred from connecting to it.
3. For each route type, assign a code number from 1 to 63. Look for the route types that are barred from accessing the same types and assign the same code number to them. If a route type is not barred from accessing any other route type, it is assigned code number 0.
4. When each route type is assigned a code number, go back to step 2 and replace the route types that the route is barred from accessing with their code number.
5. Using LD 56, create all necessary Access Restriction Tables (ARTs). Using the code number of the originating route type as the ART number, deny the necessary route type using the code numbers assigned in the previous step.
6. Assign each ART to a route in one of two methods:
7. Use LD 56 to create the Route Category Default Table (RCDT). As each route is created using LD 16, it is assigned the default ART according to route type.
8. Use LD 56 to assign to existing routes the desired ART.

The following is an example of how to configure trunk barring using the procedures listed above. This example is not reflective of the typical situation, but is only used to show the steps involved.

List all route types.

- COT
- TIE
- DID
- PAG
- DIC
- RAN – ignore because it cannot be barred.
- MDM

List route types to which the originator is barred access.

- COT is not barred from accessing any type.
- TIE is barred from accessing COT, PAG, DIC, and DID.
- DID is barred from accessing TIE, DIC, and MDM.

- PAG cannot be originator, but can be barred by other route types.
- DIC cannot be originator, but can be barred by other route types.
- MDM is barred from accessing COT, DID, PAG, and DIC.

Assign each originating route type a code number from 0 to 63.

- COT is assigned 0 (it is not barred access to any route type).
- TIE is assigned 1.
- DID is assigned 2.
- PAG is assigned 0 (this cannot be an originating route, but it can be barred by other route types).
- DIC is assigned 0 (this cannot be an originating route, but it can be barred by other route types).
- MDM is assigned 3.

Replace the route types the originator is barred from accessing with their code numbers.

- COT (ART 0) – not barred.
- TIE (ART 1) – is barred from accessing 0, and 2.
- DID (ART 2) – is barred from accessing 0, 1, and 3.
- PAG/DIC (ART 3) – not barred.
- MDM (ART 4) – bars 0, and 2.

Configure the Route Category Default Table (RCDT).

- COT 0.
- TIE 1.
- DID 2.
- OTH 0 – MDM will initially be assigned ART 0 like DIC and PAG, but can be changed using the RART prompt.

Table 109: LD 56

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TBAR	Add or change Access Restriction Table(s) (ARTs).
ART	(0)-63	Select ART to add or change. If ART table 0 is defined, no restrictions apply.
	<CR>	Return to REQ prompt.
DENY	yyy yyy	Enter ART numbers denied to Originating Trunk Connection (OTC).

Prompt	Response	Description
	ALL xALL Xyyy Xyyy	Deny all ARTs to OTC. All ART numbers allowed to OTC. Enter ART numbers allowed to OTC, or change to remove previously blocked connections.
	<CR>	Return to REQ prompt with no table being stored.

Table 110: LD 56

Prompt	Response	Description
REQ	CHG PRT	Change or Print. REQ = NEW, or OUT is disallowed for RART.
TYPE	RART	Change ART number for the route.
CUST	xx	Customer number, as defined in LD 15
ROUT	(0)-511 (0)-127	Route number.
ART	(0)-63 <CR>	ART to assign to route(s). If ART table 0 is defined, no restrictions apply. Return to REQ prompt. ART remains unchanged.

Table 111: LD 56

Prompt	Response	Description
REQ	CHG PRT	Change or Print. REQ = NEW, or OUT is disallowed for RCDT.
TYPE	RCDT	Change the route category default table.
COT	(0)-63	COT, FEX, and WAT routes are assigned the entered ART when the route is created in LD 16.
DID	(0)-63	DID routes are assigned the entered ART when the route is created in LD 16.
TIE	(0)-63	CAA, CAM, CSA, and TIE routes are assigned the entered ART when the route is created in LD 16.
OTH	(0)-63 <CR>	ADM, DIC, MDM, PAG, and RCD routes are assigned the entered ART when the route is created in LD 16. Return to the REQ prompt.

Feature operation

Barring is implemented via service change by a qualified technician. If the connection is not allowed, intercept treatment defined by the ACCD prompt in LD 15 is implemented.

Chapter 47: Trunk Failure Monitor Enhancement

Contents

This section contains information on the following topics:

Feature description

This enhancement to the Trunk Failure Monitor on page 325 feature provides a visual display on M2250 attendant consoles to indicate Direct Inward Dialing (DID)/Direct Outward Dialing (DOD)/TIE trunk line-break alarm conditions, and optionally to indicate 2.0 Mbps Digital Trunk Interface or Primary Rate Interface (PRI) Out-of-service conditions. The upper-most left key lamps on the console flash to indicate these trouble conditions.

Operating parameters

Trunk Failure Monitor (TFM) package 182 must be equipped.

This enhancement is not supported for:

- Tenant Groups attendants
- 1.5 Mbps Digital Trunk Interface (DTI), and
- Automatic Trunk Maintenance.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires Trunk Failure Monitor (TFM) package 182.

Feature implementation

Table 112: LD 15 - Configure the attendant trunk failure display.

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	FTR	Features and options
...		
- TFDR	(NO) YES	Trunk Failure Display required. Prompted with TFM package 182. Requires M2250 console.

Feature operation

The upper-most left key lamps on the console flash to indicate trouble conditions.

If the attendant is in Position Busy, Night Service, or Loop Busy state, without a call on the console, pressing the upper-most left key causes the display to show the failed trunk unit or loop number. The lamp state changes from flashing to lit. If there is more than one failed trunk or loop, the display shows them one at a time, and the lamps remain flashing until all failed trunk units or loop numbers are displayed.

When the trouble conditions have been resolved, the lamps become dark to indicate that the trunk or loop is available for normal use.

Chapter 48: Trunk Failure Monitor

Contents

This section contains information on the following topics:

Feature description

The Trunk Failure Monitor (TFM) feature detects Line Break Alarm Signals (LBAS), which are generated because of trouble conditions on Direct Inward Dialing (DID), Direct Outward Dialing (DOD), or TIE trunks, or service degraded to Out-of-service (OOS) on 2.0 Mbps Digital Trunk Interface (DTI) or Primary Rate Interface (PRI) trunks. If a line break is detected, a trunk message is printed on the maintenance TTY, and the affected trunk is rendered BUSY to stop any further seizure of the trunk during outgoing calls.

Once the line break trouble condition has been fixed, a different Line Break Alarm Signal (LBAS) is generated. The TFM feature detects this signal, prints another trunk message on the TTY indicating that the trouble condition has been corrected, and renders the repaired trunk unit IDLE for normal use.

Operating parameters

TFM is not supported by the Attendant Administration feature.

TFM is not supported on 1.5 Mbps DTI.

TFM requires the QPC730B for DID or DOD trunks, and the QPC774 for TIE trunks.

A Centralized Attendant Service (CAS) attendant can only monitor the trunks on the switch on which the attendant is located.

This feature is supported on the M2250 attendant console.

Feature interactions

Extended DID/DOD Software Support - Europe

As part of the Trunk Failure Monitor feature, the BAR/UNBAR messages are received from IPE XDID trunks (LD 15 must be configured with TFDR = YES); when a BAR message indicating a problem situation is received, a TRK501 message is printed on the TTY, the uppermost key lamps light up on the attendant console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A TRK502 message is printed on the TTY, the lamps on the attendant console are darkened, and the trunk is idled. Note that Baring Allowed (BARA) CLS must be configured on the XDID trunk for the described process to occur.

Extended Flexible Central Office Trunk Software Support

As part of the Trunk Failure Monitor feature, the BAR/UNBAR messages are received from IPE XFCOT trunks. When a BAR message indicating a problem situation is received, a trunk message is printed on the TTY, the uppermost key lamps light up on the attendant console, and the trunk is placed into BUSY state to prevent the trunk from being seized for new outgoing calls. The reception of an UNBAR message indicates that the problem situation has been cleared. A message is printed on the TTY, the lamps on the attendant console are darkened, and the seized trunk is idled. Note that BARA Class of Service must be configured on the trunk for the described processing to occur.

Feature packaging

Trunk failure Monitor (TFM) package 182.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 49: Trunk to Trunk Connection

Contents

This section contains information on the following topics:

Feature description

The Trunk to Trunk Connection feature introduces the following capabilities: transfer on ringing of external trunk across the network, transfer of one supervised outgoing external trunk to another, conference of external trunks and outgoing trunk to trunk charging. These capabilities are available on an analog (500/2500-type) telephone, proprietary telephone or an attendant console.

Transfer on Ringing of External Trunk over Network

Allows the transfer on ringing of an established external trunk call over a supervised analog network TIE trunk. If the called party does not answer within a specified time, the call will slow answer recall to the attendant on the transferring node. This capability ensures that available network resources are not occupied indefinitely.

Transfer of External Trunks

Allows the transfer of one outgoing external trunk to another trunk provided both calls are answered and both trunks have answer supervision.

As illustrated in Figure 10: Transfer on ringing of external call on page 330, Set A is on an incoming/outgoing call with Set X, an external trunk. Set A initiates a call transfer of Set X to Set B. With the Trunk to Trunk Connection feature, Set A can transfer on ringing without waiting for Set B to answer. If Set B does not answer the transferred call, the external trunk will slow answer recalls to the attendant on the transferring node.

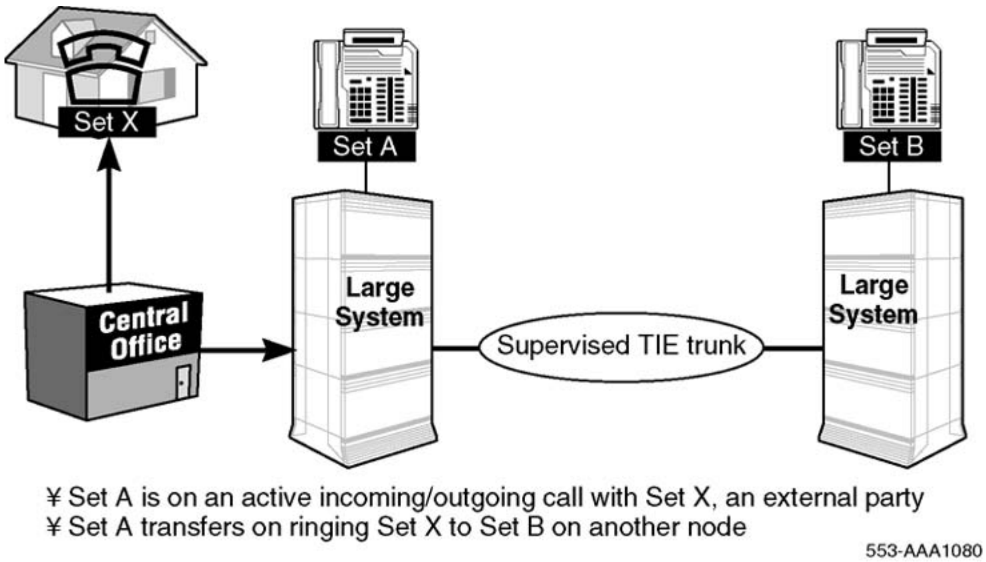


Figure 10: Transfer on ringing of external call

Conference of External Trunks

Allows external trunks to remain established in a conference call in circumstances when all external trunks involved in the call offer disconnect supervision.

Figure 11: Conference of external trunks on page 331 illustrates the Conference of External Trunk capabilities of this feature. Set A is on an established conference call with two or more external trunks, Set X and Set Y. When Set A disconnects during the conference, Set X and Set Y continue in the established call.

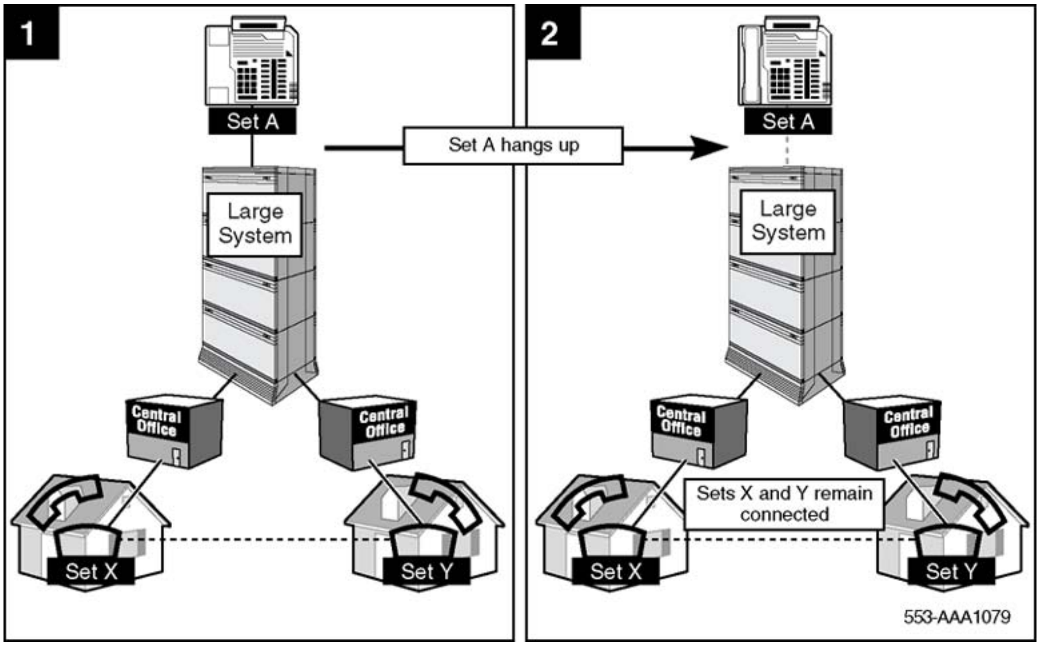


Figure 11: Conference of external trunks

Outgoing Trunk to Trunk Charging

Ensures that outstanding charging information, relevant to both outgoing calls, is contained in relevant Call Detail Recording records.

Operating parameters

Slow answer recall occurs when an external trunk is transferred on ringing across an answer supervised network TIE trunk to a telephone that does not answer. However, the resulting recall will be to an attendant on the transferring node and not to the original telephone which transferred the call.

When transferring one outgoing trunk to another, it is required that the two external calls involved are both answered prior to completing the transfer. Both external trunks involved must have both answer and disconnect supervision.

When the last internal party disconnects from a conference call, involving two or more external trunks, all external trunks must have disconnect supervision for the call to remain established. If any one of the remaining external trunks does not have disconnect supervision, all external trunks will be dropped.

No change is made to existing VNS operation.

Feature interactions

Busy Tone Detection for Japan

Busy Tone Detection for Japan does not impact Trunk to Trunk Connection. However, whichever occurs first, prevails.

Call Transfer

To transfer an external trunk on ringing across a supervised analog network TIE trunk, the external trunk and internal TIE line must have both answer and disconnect supervision, and the external call must be established. To transfer one outgoing external trunk to another, both external trunks must have answer and disconnect supervision, and both external calls must be established.

If package 131 "SUPP X11 International Basic" is equipped, to transfer one outgoing SL1 TIE trunk to another, Network Attendant Service signalling transport (NASA) must be set to YES for a corresponding DCH. Note that the Network Attendant Service feature does not need to be provisioned. However, the additional capabilities under package 131 require the NAS signalling.

Conference

Trunk to Trunk Connection allows external trunks to remain established in a call, provided that all external trunks involved have disconnect supervision. With respect to charging costs associated with a conference call, once the last telephone involved in the conference call disconnects, a search is made of all remaining trunks in the call to determine which call is established in the call for the longest period of time. This trunk is the chargeable Terminal Number (TN). This process is repeated to find the next chargeable TN.

If package 131 "SUPP X11 International Basic" is equipped, then in order for external trunks to remain established in a call after the last internal party disconnects from a conference call involving two or more external trunks, Network Attendant Service signaling transport (NASA) must be set to YES for a corresponding DCH. Note that the Network Attendant Service (NAS) feature does not need to be provisioned; however, the additional capabilities under package 131 require the NAS signaling.

Multi-Party Operations - Ringing No Answer

In a standalone environment, the RGNA prompt in the Customer Data Block will be used when an external trunk is transferred on ringing and the called party does not answer. In a network environment, the RTIM timer value in the Customer Data Block will be used for slow answer recall.

Message Registration

The last party releasing the call collects the total value of outstanding Periodic Pulse Metering (PPM) generated on outgoing trunks. If the last party is an internal 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.

Night Service

If an attendant is placed in Night Service, calls to the attendant are directed to a station with the Night DN. Recalls are not directed to the Night DN. Recalls are put in the attendant call waiting queue when in Night Service.

Night Service Enhancement

Recalls made while the attendant is in Night Service are routed to the Night DN, if the original call is an external call. In such a case, the destination party is disconnected, the internal network trunk is released and the original extended call is presented to the Night DN. If the original call is internal, recalls are put in the attendant call waiting queue when in Night Service.

Trunk Barring Trunk Group Access Restriction

Trunk Barring and Trunk Group Access Restriction takes precedence over the Trunk to Trunk Connection feature.

Feature packaging

This feature is included in base system software.

DID to TIE (DTOT) for Japan package 176 must be restricted to enable this feature.

Feature implementation

Task summary list

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

1. Table 113: LD 15 on page 334

   Modifications to Customer Data Block.
2. Table 114: LD 15 on page 334

   Modifications to Customer Data Block.

Table 113: LD 15

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	ATT_DATA	Attendant console options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
RTIM	xxx yyy zzz	Enter defined value for the Slow Answer Recall timer where: xxx = 0-(30)-378 Slow Answer Recall yy y= 0-(30)-510 Camp On Recall zzz = 0-(30)-510 Call Waiting Recall

Table 114: LD 15

Prompt	Response	Description
REQ:	CHG	Change
TYPE:	NET	Trunk and network options.
CUST		Customer number

Prompt	Response	Description
	0-99	Range for Large System and CS 1000E system.
...		
ISDN	YES	Change the Integrated Services Digital Network options.
- PSTN	NO	Public Switched Telephone Network. Limit the number of PSTNs allowed in a network connection to one PSTN. NO = Put no limit on the number of PSTN connections. YES = Limit the number of PSTN connections.
...		
DITI	YES	Allow Direct Inward Dialing to TIE connections for customer.
TRNX	YES	YES = Allow transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network. NO = Prevent transfer on ringing of an external trunk over a supervised analog network TIE trunk across private network.
EXTT	(NO)	NO = Prevent connection of supervised external trunks through either Call Transfer or Conference.
	YES	YES = Allow connection of supervised external trunks through either Call Transfer or Conference.
	<CR>	CR = Carriage return past the prompt.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 50: Trunk Traffic Reporting Enhancement

Contents

This section contains information on the following topics:

Feature description

The following modifications to trunk traffic reporting have been implemented to improve the accuracy of TFC002 traffic reports. The options are selected in the Configuration Data Block.

Traffic Period Option

Without enabling this option, trunk usage added its entire duration into the traffic period in which the disconnection occurred. If the duration was longer than 36 CCS (CCS = 100 call seconds), but less than 50 CCS, a TFS401 message was output. However, that duration was still accumulated and included in the traffic reports. If the duration was longer than or equal to 50 CCS, a TFS402 message was output. This duration was not accumulated, and was excluded from the traffic reports.

The Traffic Period Option enables the CCS to be reported in each traffic report interval. The peg count is still reported at disconnect time as per existing operation.

Note that when the Traffic Period Option is first enabled, the first traffic report may get some TFS403 messages.

Trunk Seizure Option

Without enabling this option, system traffic statistics began accumulating when a call was established. system software determined that the call was established when one of the following occurred: the End-of-Dialing (EOD) timer timed out after the last digit was dialed; the octothorpe (#) was dialed; or answer supervision was received. In some situations, customers could not match system traffic reports with their carrier reports, because many carriers start accumulating statistics when a trunk is seized.

The Trunk Seizure Option provides the ability to start accumulating statistics upon trunk seizure, rather than when the call is established.

Operating parameters

If the duration of a call is less than two to four seconds, the peg count is not accumulated. This functionality only applies when the trunk seizure option is enabled.

Due to the accumulation at trunk seizure, peg counts occur even if a call is unanswered.

Feature interactions

Automatic Call Distribution

A trunk call to an Automatic Call Distribution (ACD) DN will only be considered established once this call is answered. It is not considered established while the call is waiting in the ACD queue. Therefore, at the end of a traffic period, if a trunk call is in the ACD queue, the Traffic Period Option will not accumulate the duration for this call.

Note that when the duration is accumulated at disconnect or at the end of a traffic period after this call is answered, the total duration including the time the call was in the ACD queue is accumulated. This total duration may be longer than a single traffic period due to the time in the ACD queue and a TFS401, TFS402, or TFS403 message may be output.

Music Trunks

The Trunk Seizure Option is not supported on Music trunks.

Recorded Announcement Trunks

The Trunk Seizure Option is not supported on Recorded Announcement trunks.

Traffic Monitor

The Traffic Monitor feature outputs certain traffic data approximately every minute. The trunk usage and peg count output by the Traffic Monitor feature can be enhanced by enabling the Trunk Seizure Option. The accumulated duration and peg count of a call will begin at trunk seizure time instead of at the time the call was established.

The Traffic Monitor output that starts during the same time that the regular traffic report starts is impacted if the Traffic Period Option is enabled. With this option enabled, the duration of all currently established calls is accumulated at the end of the traffic period. Therefore, this additional duration is also accumulated in the next minute's traffic monitor output. For example, the Traffic Monitor feature and the Traffic Period Option are both enabled. Regular traffic reports are output every half hour. The difference in the accumulated duration from 10:29 to 10:30 may increase dramatically due to the additional durations accumulated for currently established calls at the end of this traffic period.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 115: LD 17 - Configure Traffic Reporting option.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	PARM	Gate opener.

Prompt	Response	Description
...		
- TPO	(NO), YES	Traffic Period Option. Enter YES to enable, NO to disable, and <CR> to keep the current value.
- TSO	(NO), YES	Trunk Seizure Option. Enter YES to enable, NO to disable, and <CR> to keep the current value.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 51: Trunk Verification from a Station

Contents

This section contains information on the following topics:

Feature description

Trunk Verification from a Station (TVS) provides the capability for a classmarked 2500-type telephone (that is, basic push-button telephone having no feature keys) to seize a particular trunk within a trunk group, receive a dial tone, and outpulse digits to complete a call to a remote maintenance site. This feature is used as part of a PC-based Network Management system to allow physical testing of each trunk in the network.

Any compatible, customer-provided PC-based circuit switched network administration and maintenance system can access the trunk to be tested by calling a remote customer-provided responder. The responder supplies the various tones needed to perform the trunk test. The PC then stores and processes the results. Once the testing is complete, the PC disconnects from the tested trunk and accesses the next trunk in the route.

To the system, the PC appears as a 2500-type telephone, which requires the capability to seize a particular trunk member within a trunk route.

Operating parameters

It is recommended that the telephone with a Trunk Verification Allowed (TVA) Class of Service also have CFW All Calls To External DN Denied (CXFD), CFW Busy Denied (FBD), and CFW No Answer Denied (FND) Classes of Service. This setup prevents any restricted telephone from accessing trunks by calling the TVA telephone and subsequently getting transferred or forwarded.

Also, it is strongly recommended that this unit not be configured with an LPA. This will prevent the unit from initiating the PBXT (test message waiting lamps) command in LD 32.

The telephone with a Trunk Verification Allowed (TVA) Class of Service should also be assigned Warning Tone Denied (WTD) Class of Service. This will prevent Attendant Busy Verification, which could impair the trunk frequency measurements that take place during a TVS call. This also prevents the trunk that this telephone has seized from being barged into by the attendant.

Trunk Verification from a Station is not applicable to B-channels on digital links.

When using the Trunk Verification feature to test network trunks, any trunk state other than an idle, such as busy, disabled or maintenance busy, an overflow tone is returned.

Feature interactions

The environment in which the TVS feature will be invoked is a machine environment. That is, the user of the 2500-type telephone with this feature will usually be a PC-based maintenance system. Therefore, minimal interaction exists with other features.

When the 2500-type telephone with a TVA Class of Service makes a TVS call, any Trunk Group Access Restrictions/Trunk Access Restriction Groups (TGAR/TARG) defined in the system are removed for this call.

When a trunk group is busied out by an attendant console, access to that trunk group is not allowed with the TVS feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 116: LD 10 - Allow or deny Trunk Verification from a 2500 telephone.

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	(TVD) TVA DTN	(Deny) allow TVS. Digitone service is required for 2500 telephones.

Feature operation

To verify that a trunk is working properly (from a 2500 telephone with TVA Class of Service), follow these steps:

1. Lift the handset.
2. Dial SPRE + 70 + ACOD + mmm

where:

SPRE	is the special function access prefix
70	is the special access code for the TVS feature
ACOD	is the access code of the trunk group to be tested, and
mmm	is the number of the trunk member that is to be seized; mmm must be three digits (for example, 001).

Chapter 52: Uninterrupted Line Connections

Contents

This section contains information on the following topics:

Feature description

Uninterrupted Line Connections are connections assigned Warning Tone Denied (WTD) Class of Service. The feature prohibits the imposition of any Camp On or intrusion tones on that line.

This feature is recommended for modem or data lines.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Barge-In, Attendant Busy Verify, Override

These features cannot be applied to stations with a WTD Class of Service.

Camp-On

A call can be camped on to a station with a WTD Class of Service, but tone is not provided.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 117: LD 10 on page 347

   Assign Warning Tone Allowed for analog (500/2500-type) telephones.
2. Table 118: LD 11 on page 347

   Assign Warning Tone Allowed for Meridian 1 proprietary telephones.
3. Table 119: LD 14 on page 347

   Assign Warning Tone Allowed for trunks.

Table 117: 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	(WTA) WTD	Warning tone (allowed) denied.

Table 118: 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	(WTA) WTD	Warning tone (allowed) denied.

Table 119: LD 14

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Trunk type, where: aaa = ADM, AID, ATVN, AWR, CAA, CAM, COT, CSA, DIC, DID, FEX, ISA, MDM, MUS, PAG, RAN, RCD, RLM, RLR, TIE, or WAT.
TN	l s c u	Terminal Number For Large Systems
CLS	(WTA) WTD	Warning tone (allowed) denied.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 53: United Kingdom Analog Hardware Support

Contents

This section contains information on the following topics:

Feature description

The United Kingdom Analog Hardware Support feature provides the following capabilities:

- UK Analog Trunk Enhancements
- UK Transmission Plans

UK Analog Trunk Enhancements

Software changes have been implemented for the following hardware packs, in order to comply with UK standards:

- XDID (Extended DID trunk card)
- XCOT (Extended Central Office trunk card)
- XTD (Extended Tone Detector card), and
- XFEM (Extended Flexible E&M trunk card).

XDID

Situation	Solution
A DID trunk is not available for a new call.	A backward signal is sent to the Public Switched Telephone Network.
A short line and long line DID trunk requires support.	A 2dB Short Line (SHL) and Long Line (LOL) pad matrix have been defined.

XCOT

Situation	Solution
Support the following types of disconnect signaling required for Central Office trunks: • Earth Signaling (Ground Start), • Loop Calling (Disconnect Clearing), and • Loop Calling (Guarded Release) signaling.	The appropriate disconnect sequences have been programmed.
For Periodic Pulse Metering (PPM), an option is required to default to a meter pulse frequency of 50 Hz (the XCOT pack for the UK can only accept this value).	In the Route Data Block, if the PPM frequency is not prompted, the value will default to 50 Hz.
For Periodic Pulse Metering, the counting of buffered and unbuffered pulses.	The software has been modified to support both buffered and unbuffered PPM pulses.
A time-configurable detector is required to monitor the disconnection of loop trunks, disconnect clear trunks, and release guard trunks.	The Loop Calling Timer (LCT), with a configurable range of 128-32640 milliseconds, has been introduced in the Route Data Block.
UK ringing must be recognized.	To recognize UK ringing, the default value of the ring validation timer has been changed from 512 milliseconds to 256 milliseconds.
UK COT with Earth Signalling (Ground Start) or Loop Calling Disconnect Clearing provides hardware answer supervision.	The software has been modified to support answer supervision for both Earth Signalling (Ground Start) and Loop Calling Disconnect Clearing. Prompt SUPN appears for both types of signaling in LD 14. Answer supervision is not provided for Loop Calling Guarded Release.

XTD

The XTD pack can be configured, on a per-call basis, for either Dual-tone Multifrequency (DTMF) or Dial Tone Detection (DTD) signaling.

XFEM

The XFEM pack supports recorded announcement trunks, paging trunks, and music trunks, two-wire E&M, four-wire E&M, and 2280 Hz TIE trunks.

UK Digital Transmission Plans

Software changes have been implemented in order to comply with UK digital transmission plans for the following:

- Digital trunks, and
- Meridian Modular telephones.

Digital trunks

Situation	Solution
The transmission parameter values for digital trunks must be fixed.	The transmission parameter values for digital trunks are automatically downloaded, based on a zero default value.

Meridian Modular telephones

Situation	Solution
The transmission parameter values must be fixed and automatically downloaded, on a per-system basis.	The software has been changed to prevent transmission parameter prompts from appearing. The transmission parameters will be fixed for the UK, and will be downloaded on a per-system basis.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 120: LD 13 on page 352

   Configure the DTD/DTR data block.
2. Table 121: LD 14 on page 353

   Configure UK trunks, their associated signaling and transmission option.
3. Table 122: LD 16 on page 353

   Define the Loop Calling Detection Timer, and how the ground signal from a Recorded Announcement (RAN) machine should be interpreted for XFEM cards.

Table 120: LD 13

Prompt	Response	Description
...		

Prompt	Response	Description
TYPE	XTD	Extended Dial Tone Detector and Digitone Receiver data block.
...		
XTDT	(0)-7	Extended Tone Detector Table Number, prompted when type = XTD. If a table other than 0 is entered, it must exist in LD 97.
- DTO	(NO) YES	Dial Tone Detection Only. (NO) = Do not disable DTR detection. YES = Disable DTR detection, only perform dial tone detection.

Table 121: LD 14

Prompt	Response	Description
...		
XTRK	XFEM XDID XCOT	Extended Flexible E&M trunk card. Extended DID trunk card. Extended CO trunk card.
...		
SIGL		Trunk signaling.
	LDC LGR	Loop calling, disconnect clear. Accepted when TYPE = COT and UK package is equipped. Loop calling, guarded release. Accepted when TYPE = COT and UK package is equipped.
...		
CLS	(SHL) LOL	(Short line) Long line Class of Service.
	NTC TRC VNL	Transmission Class of Service, where: NTC = Non-transmission Compensated TRC = Transmission Compensated, and VNL = Via Net Loss.
		For E&M4 Wire and AC15 defined on XFEM trunks, NTC is used for circuit switched network-to-PSTN Link connections, while VNL is used for circuit switched network-to-circuit switched network TIE connections. SHL replaces TRC and LOL replaces NTC and VNL for XDID and XCOT trunks in Phase 7C and later.

Table 122: LD 16

Prompt	Response	Description
...		
TIMR	LCT 0-128-1280	Loop Calling Detection Timer in milliseconds.

Prompt	Response	Description
		Default for COT trunks = 128. Default for all other trunks = 256.
...		
GRD		Determines how the ground signal from a RAN machine should be interpreted for XFEM cards.
	(PLAY)	The ground signal from the RAN machine indicates that the machine is playing.
	IDLE	The ground signal from the RAN machine indicates that the machine is idle.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 54: User Selectable Call Redirection

Contents

This section contains information on the following topics:

Feature description

User Selectable Call Redirection (USCR), enhances the implementation of several existing features. First, it enables the user to modify DNs at the telephone for the following redirections:

- Flexible Call Forward No Answer DN (FDN)
- Hunt DN (HUNT)
- External Flexible Call Forward No Answer DN (EFD), and
- External Hunt DN (EHT).

The Station Control Password feature must be active, with passwords defined in LD 15, for the user to change these redirection DNs.

Second, it expands the number of selectable Ringing Cycle Options (RCOs) for Flexible Call Forward No Answer (CFNA) from one to three.

User assignment of redirection DNs

USCR permits the user to modify any of the redirection DN's for FDN, HUNT, EFD, and EHT from a rotary, push-button, or digital telephone.

Depending on the type of telephone, there are three ways to access this feature: using a Special Service Prefix Code (SPRE 9915), a Flexible Feature Code (FFC), or the User Selectable Redirection (USR) key.

The user can also change the RCO from a telephone after accessing USCR. For security reasons, the user must enter the Station Control Password (SCPW) before changing the redirection DNs or the RCO.

Ringing Cycle Options (RCOs) for CFNA

The original implementation of Call Forward No Answer provided a single option (CFNA in LD 15) that defined the number of normal ringing cycles before CFNA treatment. The value could be in the range of 1-15, with a default of 4. This value determined how many times the telephone rang before CFNA treatment was initiated.

The CFNA prompt is now replaced with prompts CFN0, CFN1, and CFN2, each of whose value can be in the range of 1-15, with a default of 4. The number of distinctive ringing cycles for CFNA is also expanded. The DFNA prompt in LD 15 is replaced with DFN0, DFN1, and DFN2, with the same value range and default.

Additionally, the Ringing Cycle Option (RCO) prompt appears in LDs 10 and 11 for each telephone. Its value, in the range of 0-2, is a pointer to the CFNx and DFNx entries in the Customer Data Block. The following chart explains the relationship of the RCO value and the CFNx and DFNx entries in the Customer Data Block.

Operating parameters

To assign or print the RCO for a telephone requires that it have the Flexible Call Forward No Answer Allowed (FNA) Class of Service or Message Waiting Allowed (MWA) Class of Service.

The user's telephone must have User Selectable Redirection Allowed (USRA) Class of Service and a Station Control Password (SCPW). The user must enter the correct password to access USCR.

Table 123: Relationship between RCO value and CFNx, DFNx contents

An RCO value (per telephone) of	Selects these CFNA and DFNA entries (with sample contents shown)	And has this effect
0	CFN0 (Default value of 4) DFN0 (Value set to 2)	CFNA treatment after four rings CFNA treatment after two distinctive rings
1	CFN1 (Value set to 6) DFN1 (Value set to 5)	CFNA treatment after six rings CFNA treatment after five distinctive rings
2	CFN2 (Value set to 3) DFN2 (Default value of 4)	CFNA treatment after three rings CFNA treatment after four distinctive rings

Basic Rate Interface (BRI) telephones do not support USCR because they cannot access SPRE or FFC, and have no feature keys. Therefore, BRI telephones will always use the entries for CFN0 and DFN0.

The user cannot use USCR to initially configure call redirection features. The features must be equipped, and the initial call redirection DNs must be established, via a service change.

This feature cannot be used remotely. A user can only change redirection DNs or the RCO for the telephone being used to access USCR.

Feature interactions

Automatic Call Distribution

An Automatic Call Distribution (ACD) DN cannot be stored as a redirection DN unless the ACD queue is defined as a Message Center.

Attendant Administration

Attendant Administration does not support assigning the USR key, RCO, or USRA/USRD Class of Service.

Autodial

USCR does not support Autodial; Autodial cannot be used to dial all or part of the digits for USCR programming.

Call Forward All Calls

When CFW redirects a call from telephone A to telephone B, and telephone B does not answer, the RCO of telephone B determines how long it rings. After the designated number of rings, the FDN of telephone A redirects the call.

Call Forward by Call Type

USCR enables a user to assign EFD from the telephone.

Call Forward No Answer Flexible Call Forward No Answer

The single parameters previously used to define normal ringing cycles (CFNA) and distinctive ringing cycles (DFNA) are expanded to three (CFN0-2 and DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific CFNA and DFNA entries for each telephone.

Call Forward No Answer, Second Level

The number of ringing cycles before Second Level Call Forward No Answer (SFA) is determined by the RCO for the ringing DN, as with CFNA.

Call Redirection by Time of Day

User Selectable Call Redirection is not supported by Call Redirection by Time of Day.

Dial Access to Features and Services

The 9915 feature code accesses USCR from an analog (500/2500-type) telephone or a Meridian 1 proprietary telephone. The user dials this code after dialing the SPRE.

Directory Number Delayed Ringing

With User Selectable Call Redirection (USCR) a user can change the number of CFNA/DFNA ringing cycles. If the user changes the CFNA/DFNA value so that CFNA takes place before the Directory Number Delayed Ringing timer runs out, none of the SCN/MCN keys will receive an audible notification.

Distinctive/New Distinctive Ringing

The single parameter previously used to define distinctive ringing cycles (DFNA) is expanded to three (DFN0-2), with the Ringing Cycle Options (RCO) parameter used to select the specific DFNA entry for each telephone.

DPNSS1 Diversion

The User Selectable Call Redirection feature triggers Diversion Validation. If the numbering plan is DPNSS1 then diversion occurs. Numbering plan routes are checked to determine if redirection DN's are through DPNSS1 on a first choice route basis. If the number plan is not a DN through DPNSS1, then User Selectable Call Redirection works as usual.

Enhanced Hot Line Flexible Hot Line

An analog (500/2500-type) telephone with a Hot Line feature cannot use User Selectable Call Redirection, because it cannot access any features through SPRE or FFC.

Hunting

User Selectable Call Redirection permits a user to change the HUNT DN or EHT from a telephone. An attendant DN is only allowed for HUNT and EHT if the customer has the attendant defined as a message center (LD 15 – MATT=YES).

Message Center (MC) and Message Waiting

USCR affects the number of times the DN rings before the call is forwarded to the Message Center. The RCO in the Terminal Number (TN) block of the Multiple Appearance Redirection Prime (MARP) for the called DN determines the number of times the DN rings.

Multiple Appearance Redirection Prime (MARP)

When a Multiple Appearance DN is rung, the determination of the number of ringing cycles for CFNA depends on the value of the MARP prompt in LD 17. If the value is YES, the number of ringing cycles is determined by the RCO number of the DN that is classified as a MARP TN.

If the DN is a Multiple Appearance DN (MADN), the RCO values in the other TN blocks for that DN are ignored.

If the MARP value is NO, the RCO is taken from the first TN in the DN block with a primary appearance of the DN. If there is none, the last TN in the DN block is used.

Pretranslation

If Pretranslation (package 92) is enabled, the digits entered as the redirection DN are pretranslated before they are stored. Note that no Pretranslation occurs when the redirection DNs are used in such call processing features as Hunting or CFNA, eliminating the possibility that the redirection DN is pretranslated twice.

Short Hunting

USCR does not support changing the HUNT or EHT for a telephone with Short Hunt enabled. USCR also does not support entering 000 from a telephone as the HUNT.

Speed Call

Speed Call is not supported by USCR.

Feature packaging

Flexible Feature Codes (FFC) package 139 is a prerequisite for the user activation part of this feature because it provides for the Station Control Password.

Feature implementation

Task summary list

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

1. Table 124: LD 15 on page 361

Configure USCR in the Customer Data Block.

2. Table 125: LD 10 on page 361

   Configure USCR for analog (500/2500-type) telephones.

3. Table 126: LD 11 on page 362

   Configure USCR for Meridian 1 proprietary telephones.

4. Table 127: LD 57 on page 362

   Configure USCR Flexible Feature Codes.

Table 124: LD 15

Prompt	Response	Description
REQ:	NEW CHG	ADD, or change.
TYPE:	RDR	Call Redirection
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- CFN0	1-(4)-15	Number of normal rings for CFNA, Option 0.
- CFN1	1-(4)-15	Number of normal rings for CFNA, Option 1.
- CFN2	1-(4)-15	Number of normal rings for CFNA, Option 2.
- DFN0	1-(4)-15	Number of distinctive rings for DFNA, Option 0.
- DFN1	1-(4)-15	Number of distinctive rings for DFNA, Option 1.
- DFN2	1-(4)-15	Number of distinctive rings for DFNA, Option 2.
TYPE	FFC	Gate opener.
...		
- SCPL	(0-8	Length of Station Control Password. If 0 = password disabled, USCR cannot be used.

Table 125: LD 10

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	500	Telephone type.
RCO	(0) 1 2	Ringing Cycle Option for CFNA, in the range of 0-2, with a default of 0.
SCPW	xxx...xx	Station Control Password.
CLS	(USRD) USRA	User Selectable Redirection Class of Service (permitting SPRE and FFC access) (denied) allowed.

Prompt	Response	Description
The technician can use easy change to change the RCO and USRA/USRD CLS. At the ITEM prompt, type RCO <value> where the value is 0-2.		

Table 126: LD 11

Prompt	Response	Description
REQ:	NEW CHG	Add, or change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
RCO	(0) 1 2	Ringing Cycle Option for CFNA, in the range of 0-2, with a default of 0.
SCPW	xxx...xx	Station Control Password.
CLS	(USRD) USRA	User Selectable Redirection Class of Service (permitting SPRE, FFC, and USR key access) (denied) allowed.
KEY	xx USR	Key number of the USR key.
The technician can use easy change to change the RCO and USRA/USRD CLS. At the ITEM prompt, type RCO <value> where the value is 0-2.		

Table 127: LD 57

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
CUST	xx	Customer number, as defined in LD 15
CODE	USCR ALL	Prompt for USCR FFC, or all FFC code types.
USCR	xxxxxxx yyyyyyy <CR>	USCR FFC (1-7 digits). Define additional FFC codes, as needed. Ends the entry of FFC codes.

Feature operation

As a prerequisite to accessing the feature, the conditions shown in Table 128: Requirements for accessing USCR on page 362 must be met for the selected access method.

Table 128: Requirements for accessing USCR

Requirement	Access Method		
	USR Key	SPRE	FFC
FFC package equipped	Yes	Yes	Yes

Requirement	Access Method		
	USR Key	SPRE	FFC
SCPL is defined (>0)	Yes	Yes	Yes
SCPW is defined	Yes	Yes	Yes
Telephone has USR key	Yes	No	No
USRA Class of Service defined	Yes	Yes	Yes
SPRE defined	No	Yes	Yes
USCR FFC defined	No	No	Yes

To assign/query a redirection DN using SPRE:

- Take the telephone off-hook, or press the DN key on a digital telephone.
- Enter the SPRE.
- Enter the USCR feature access code (9915).
- Enter the Station Control Password.
- Enter the USCR option code, as shown in Table 129: USCR option codes on page 363.

Table 129: USCR option codes

Code	Used to assign
1	FDN redirection DN
2	HUNT redirection DN
3	EFD redirection DN
4	EHT redirection DN
5	RCO

- Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
- Place telephone on-hook, or press the Rls key on a Meridian 1 proprietary telephone.

To assign or query a redirection DN using the USR key:

- Press the dark USR key.
- Enter the Station Control Password.
- Enter the USCR option code from Table 129: USCR option codes on page 363.

- Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
- Press the USR key again.

To assign or query a redirection DN using an FFC:

- Take the telephone off-hook, or press the DN key on a Meridian 1 proprietary telephone.
- Enter the USCR FFC.
- Enter the Station Control Password.
- Enter the USCR option code, as shown in Table 129: USCR option codes on page 363.
- Enter the new RCO if assigning the RCO; enter the redirection DN if assigning the DN.
- Place telephone on-hook, or press the Rls key on a Meridian 1 proprietary telephone.

Chapter 55: Variable Guard Timing

Contents

This section contains information on the following topics:

Feature description

The guard timing capability for a trunk prevents outgoing calls from reseizing trunks for a specified time after disconnection, thereby protecting trunks against glare conditions. This feature allows the customer to specify one guard timing interval for incoming call disconnection and one guard timing interval for outgoing call disconnection.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 130: LD 16 - Configure Guard Timers for trunk.

Prompt	Response	Description
...		
CFWR	(NO) YES	CFW restriction (not allowed) allowed.
IDOP	(NO) YES	Respond YES to allow the trunk CDR for internal calls to identify the originating station instead of the forwarding station.
TIMR	GTI 128-(896)-32640	Incoming Guard timer.
TIMR	GTO 128-(896)-32640	Outgoing Guard timer.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 56: Virtual Office for M3900

Feature description

For information about M3900 (Single Site) Virtual Office, see Features and Services Fundamentals — Book 4 of 6 (I to M).

Chapter 57: Virtual Office login/logout for Multiple Line Appearance

Contents

This section contains information about the following topics:

Feature description

Use this feature for Virtual Office login or logout if Multiple Line Appearance exists on an IP Phone and one of the line keys is busy. The line key is busy if a call exists on a Multiple Appearance Directory Number (MADN) that another appearance originates or terminates.

Operating parameters

The operations of this feature depend on the direction of the Virtual Office action.

Virtual Office login to an IP Phone with multiple line appearance

IP Phone A is configured with Virtual Office Login Allowed (VOLA) Class of Service (CLS). IP Phone B is configured with Virtual Office User Allowed (VOUA) CLS.

IP Phone A tries to perform a Virtual Office login to IP Phone B. IP Phone B has a MADN configured as Single Call Ringing/Non-ringing (SCR/SCN). Another appearance of the same DN exists on another IP Phone and is in an active call; IP Phone B has no active calls.

The Virtual Office login from IP Phone A to IP Phone B succeeds if IP Phone A uses another DN on IP Phone B as the user ID.

Virtual Office logout from an IP Phone with Multiple Line Appearance

IP Phone A is configured with VOLA CLS. IP Phone B is configured with VOUA CLS.

IP Phone B is in the Virtual Office logout state. The Terminal Number (TN) of IP Phone B uses a MADN configured as SCR/SCN. Another appearance of the same DN exists on another IP Phone and is in an active call; IP Phone B has no active calls.

IP Phone A can log out from IP Phone B and IP Phone B returns to the home TN.

Virtual Office login from an IP Phone with Multiple Line Appearance

IP Phone C with the VOLA CLS uses a MADN configured as SCR/SCN. Another appearance of the same DN exists on another IP Phone and is in an active call; IP Phone C has no active calls. IP Phone C can perform a Virtual Office login to another IP Phone with the VOUA CLS.

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 Virtual Office Login/Logout for Multiple Line Appearance on an IP Phone.

Configuring VOLA and VOUA CLS

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

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

   The **Phone Details** Web page appears.
5. Scroll to the Features section.
6. For VOLA, select **Allowed** from the list.
7. For VOUA, select **Allowed** from the list.

   See Figure 12: VOLA and VOUA on page 371.

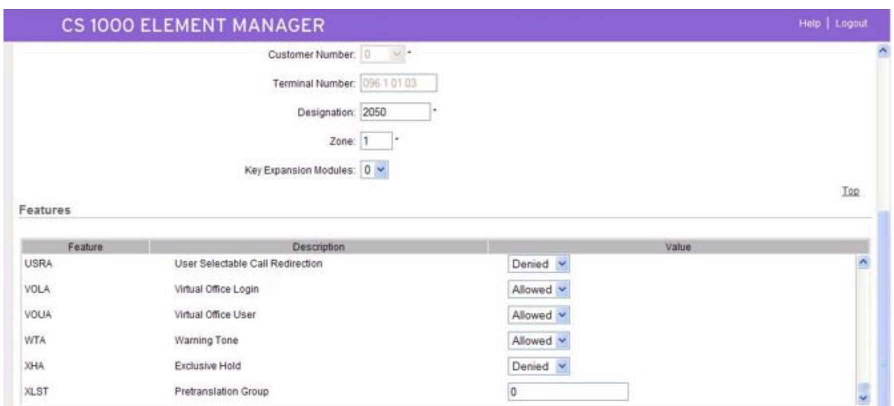


Figure 12: VOLA and VOUA

Feature operation

The home office IP Phone TN must have Virtual Office login to this TN using a remote phone (VOUA) enabled.

This means that if you want to use Virtual Office to log into your home office IP Phone from a remote IP Phone, then your home office IP Phone must have VOUA (Virtual Office User Allowed) configured.

Chapter 58: Virtual Office administrator logout

Contents

This section contains information about the following topics:

Feature description

Use this feature to locate idle Virtual Office IP Phones and log them out, as necessary. You can force logout for all IP Phones or a specific IP Phone idle for more than the configured Idle Time.

You can search for Virtual Office logged-in IP Phones based on idle time criteria in the Search For Phones Web page in Element Manager, and then log them out. You must have LD 117 permission to use this feature.

You can search for IP Phones that are idle for greater than or equal to a specified time.

Operating parameters

This feature provides the following functionality:

- logout all Default Virtual Office Logout Allowed (DVLA) logged-in idle IP Phones, which are idle for more than Idle Time minutes (if specified).
- logout a specific DVLA IP Phone if it is logged in and idle.
- parse specified file from /e/temp/ directory on the Call Server and log out all DVLA IP Phones for which Terminal Numbers (TN) exist in the file.

  The file must contain only the TN, in string format, (for example, 096 0 00 30), on each line.
- provide information (TN, Prime DN, Idle Time) about logged-in DVLA IP Phones, which are idle for more than the Idle Time minutes (if specified). This feature cannot provide more than 1000 records simultaneously. If the system collects more information, an error message prints.

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 list is a summary of the tasks in this section:

1. Searching with Idle Time on page 375
2. Advanced Searching with Idle Time on page 375
3. Logging out a Virtual Office IP Phone on page 376

To search for an Virtual Office IP Phone based on the Idle time criteria, use the following procedure:

Searching with Idle Time

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. In the Criteria list, select**Idle Time (>=)**.
4. Type the required value in the Value field.

   Important:

   You must type a value between 1 and 1440 (minutes) as the Idle Time. If you type an invalid value, the search does not complete.
5. Click **Search**.

   The page refreshes to show the results that match the search criteria.

   Important:

   If the IP Phone data is not synchronized with the Call Server, then the search returns partial results. Click the log file link to view the list of IP Phones not present in the database.

To perform an advanced search for a Virtual Office IP Phone with Idle Time, use the following procedure:

Advanced Searching with Idle Time

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. Click **Advanced**.

   The Advanced Search Web page appears.
4. In the Field list, select**IDLE TIME - Idle Time**.
5. Type the required value.

   You can use other criteria like AND and OR logic operators, and different field values.
6. Click **Search**.

   The page refreshes to show the results that match the search criteria.

To logout a Virtual Office IP Phone, use the following procedure:

Logging out a Virtual Office IP Phone

1. Log on to Element Manager with a valid user account.
2. Search for a Virtual Office IP Phone using the Searching with Idle Time on page 375 procedure or Advanced Searching with Idle Time on page 375 procedure.
3. Select the check box that corresponds to the IP Phone to log out.

   Logout from multiple IP Phones by selecting the corresponding check boxes.
4. From the More list, elect **Logout**.

   The system prompts for confirmation to log out from the selected IP Phone.
5. Click **OK**.

   The page refreshes to show the Search for Phones Web page with the selected IP Phones removed from the search result list.

Feature operation

Use this feature to log out only idle DVLA IP Phones.

A warning message appears on the IP Phone for one minute before logout. The IP Phone user can press Yes to accept the logout or press No to cancel it.

Chapter 59: Virtual Office logout during midnight routines

Contents

This section contains information about the following topics:

Feature description

Use this feature to configure automatic logout during midnight routines for inactive IP Phones with a Class of Service (CLS) of Default Virtual Office Logout Allowed (DVLA). An IP Phone with a DVLA CLS is inactive if the IP Phone user does not press a key for a configured period of time.

Operating parameters

If you enable this feature, IP Phones with a DVLA CLS unregister from the Terminal Proxy Server (TPS) during midnight routines if the IP Phones are inactive longer than the configured Idle Time .

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

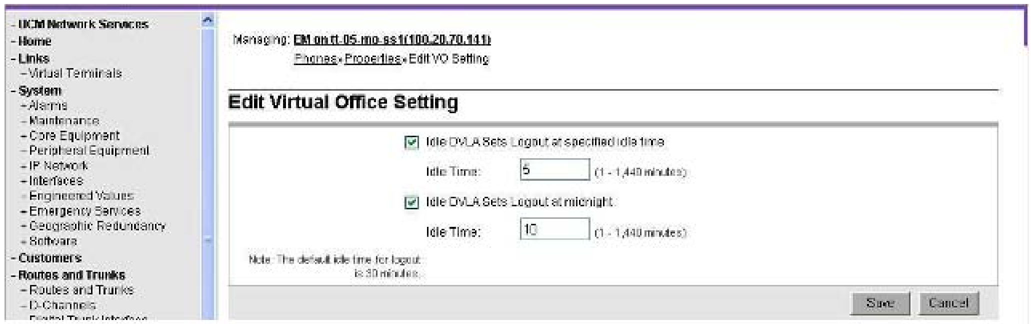
Use Element Manager to configure Virtual Office logout during midnight routines.

Configuring Virtual Office logout during midnight routines in Element Manager

1. Log on to Element Manager with a valid user account.
2. In the Navigator pane, select **Phones**, **Properties**.
3. In the Virtual Office Setting pane, click Retrieve to retrieve the Virtual Office Setting from the Call Server.

4. Click **Configure** to configure the automatic logout settings of inactive idle DVLA IP Phones.

   The Edit Virtual Office Setting Web page appears and shows the current configuration.



5. Select the check box for **Idle DVLA Sets Logout at midnight**.

   The Idle Time field appears.

6. Type the idle time in the Idle Time field.

   Important:

   The idle time must be a value between 1 and 1440 minutes. The default is 30 minutes.

7. Click **Save**.

   The page refreshes and the Properties Web page appears.

Feature operation

The system disables Virtual Office logout during midnight routines by default. You can configure Idle Time Interval to a value from 1 to 1440 minutes. The default Idle Time interval is 30 minutes.

Chapter 60: Virtual Office logout on IDLE condition

Contents

This section contains information about the following topics:

Feature description

Use this feature to configure a rule for automatic logout of idle IP Phones with a Class of Service (CLS) of Default Virtual Office Logout Allowed (DVLA). If the Virtual Office IP Phone remains in the IDLE condition for a specific time interval, the IP Phone automatically logs out. The IP Phone user receives a warning message with the option to cancel the rule and restart the IDLE timer.

Operating parameters

If you enable this feature, DLVA IP Phones which are in the IDLE condition for a specific time interval automatically log out. The IP Phone displays a warning message with the option for the IP Phone user to cancel the logout and restart the IDLE timer.

Virtual Office logout on IDLE condition is disabled by default. You can configure Idle Time Interval with a value from 1 to 1440 minutes. The default Idle Time Interval is 30 minutes.

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 Virtual Office logout on IDLE condition.

Configuring Virtual Office logout on IDLE condition in Element Manager

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

   The Edit Virtual Office Setting Web page appears and shows the current configuration.
4. Select the check box for **Logout a DVLA Set after the specified idle time has elapsed**.

   The Idle Time field appears.
5. Type the idle time in the Idle Time field.

   Important:

   The idle time must be a value between 1 and 1440 minutes.
6. Click **Save**.

   The page refreshes and the Properties Web page appears.

Feature operation

If this feature is enabled and the DVLA IP Phone is in the IDLE condition longer than the configured Idle Time interval, the message `Logout phone now?` appears on the IP Phone. The IP Phone user can select Yes or No. If the IP Phone user selects Yes, the IP Phone logs out. If the IP Phone user selects No, the Idle Time interval restarts and the IP Phone does not log out of Virtual Office. If the IP Phone user does not select a response, logout occurs.

Chapter 61: Virtual Office-only IP Phones

Contents

This section contains information about the following topics:

Feature description

Use this feature to configure Virtual Office-only IP Phones. Virtual Office-only IP Phones do not use an assigned Directory Number (DN) or consume a Terminal Number (TN) license. By default, these IP Phones are in the Virtual Office logout state. Use these IP Phones for Virtual Office login only.

Operating parameters

Configure the Default Virtual Office Logout Allowed (DVLA) Class of Service (CLS) for the IP Phone.

The IP Phone registers with the Call Server by using a TN. If the TN has a DVLA CLS, the IP Phone registers in the Virtual Office logout state.

The IP Phone is in the Default Virtual Office Logout Denied (DVLD) CLS by default and the IP Phone registers in the Virtual Office logout state.

Feature interactions

CS 1000 Branch Survivability

Normally, the IP Phone calls are directed to the main site from the branch site except for the DVLA class of service. Therefore, the IP phones for this class of service remains registered to the branch site while all other services registers to the main site.

It is recommended that you configure the phones with DVLA class of service in a CS 1000 remote branch to register with the main site. If you require the branch survivability function, use the network wide office without DVLA configured on the IP Phones.

An alternative configuration is to use S1 and S2 registration option in the phones. Point S1 to the main site and S2 to the branch site. In this configuration, the IP phones registers to the branch site for survivability. However, the phones will remain registered on the branch site until you reboot the IP phones manually. In this configuration, if you restore the main site after the outage, the resources at the branch registers again with the main site and these will be no longer available to phones with DVLA class of service. Therefore, you must reboot the phones with DVLA class of service as soon as the main site is restored. This limits the potential interruption of resource availability.

Feature packaging

This feature is included in base system software.

Feature implementation

For a Virtual Office-only IP Phone, use Element Manager to configure DVLA CLS .

Configuring DVLA 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 DVL (Default Virtual Office Logout), select **Allowed** from the list.

   See Figure 13: DVL on Phone Details Web page on page 387.

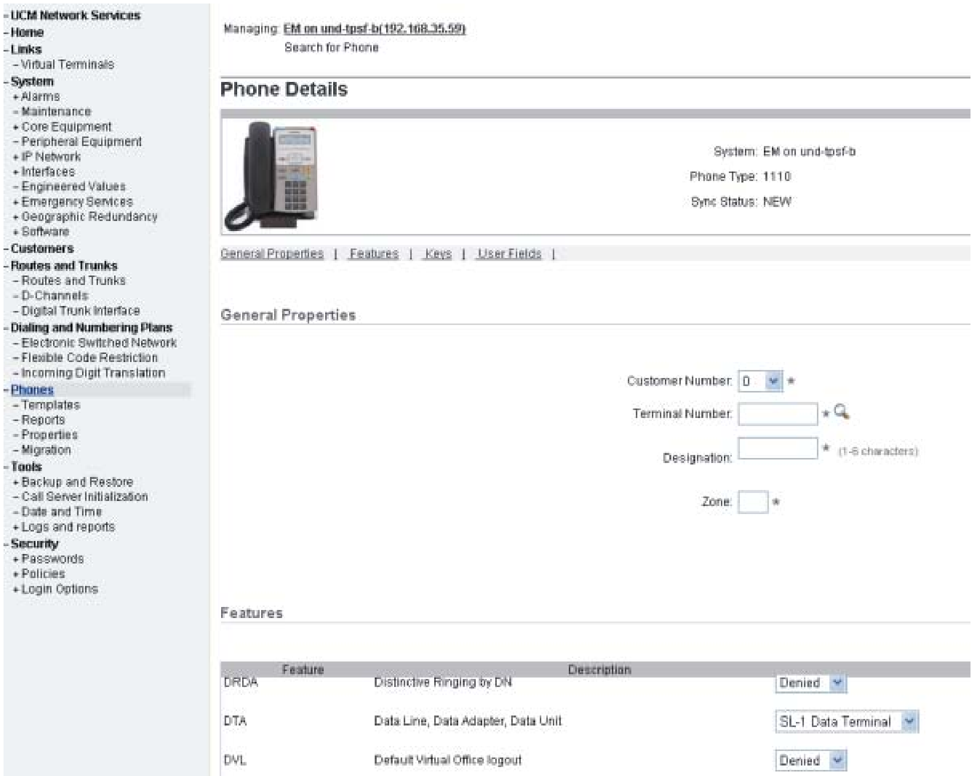


Figure 13: DVL on Phone Details Web page

Feature operation

Use the DVLA CLS IP Phone to log in to Virtual Office.

Chapter 62: Voice Call

Contents

This section contains information on the following topics:

Feature description

Voice Call allows you to talk through the speaker of a Meridian digital telephone from another Meridian digital telephone. The called party does not have to lift the handset to hear you. For a two-way conversation, the called party must lift the handset or activate Handsfree, unless Handsfree Voice Call is enabled.

If the called telephone is busy on another DN, the caller hears continuous ringing. The called party hears a single beep and the Voice Call DN key flashes. If the telephone is busy on the Voice Call DN, the caller hears a busy tone. A fast busy tone may indicate that the Voice Call DN is no longer available (it may not be a Single Appearance DN).

Handsfree Voice Call

Handsfree Voice Call is a system feature that can be used with such telephones as the M2317 and M2616.

Handsfree Voice Call provides the option of configuring VCC/DIG (with voice option) to be answered in either Handsfree mode or loudspeaker only mode. Calls answered in Handsfree (HVA) mode establish a two-way voice path, while those answered in loudspeaker only (HVD) mode establish a one-way voice path from the calling telephone to the destination telephone.

Operating parameters

Both telephones must be Meridian digital telephones.

The Voice Call DN must be single appearance.

Handsfree Voice Call allowed/denied is configured at the system level and can be used with only digital telephones that have Handsfree capabilities (such as the M2317 and M2616). It requires Handsfree Allowed/HFA Class of Service on the destination telephone, which is configured at the telephone level. Basic Rate Interface (BRI) telephones do not support the Handsfree feature.

Feature interactions

Auto Answer Back

This feature is not affected by the Handsfree Voice Call feature.

Automatic Line Selection

This feature is not selected by automatic Outgoing Line Selection. It is selected for Incoming Ringing and Non-Ringing Line Selection.

Call Party Name Display

The telephone originating a Voice Call displays the called DNs Call Party Name Display. The called telephone shows the caller's DN and name on its display.

Display of Calling Party Denied

Display information on telephones involved in a Voice Call is based on the individual Class of Service of each telephone.

Flexible Feature Code Boss Secretarial Filtering

A call to a Voice Call key on a boss telephone with filtering active is not filtered to the secretary telephone.

Flexible Voice/Data Terminal Number

If a dynamic TN has a single appearance DN key that terminates on a Voice Call (VCC) key, the called party hears a single beep if occupied on another DN. However, if the called party is a dynamic TN in data mode, the DN key lamp flashes. A beep is not provided.

Hot Line

The terminating DN of a Voice Call arrangement may be the incoming DN of a two-way Hot Line.

When engineering call-modification paths (such as Hunting and Call Forward No Answer), the Hot Line Restriction option will cancel the normal call-modification operation for internal non-Hot Line calls.

Manual Signaling

The same DN can be used for both Voice Call and Manual Signaling (Buzz) as long as it remains a Single Appearance DN.

Multiple Appearance DNs

If a Voice Call DN is added to a second telephone, the DN becomes a Multiple Appearance DN (MADN). Voice Call no longer works on that DN and fast busy tone is returned.

On Hold on Loudspeaker

It is possible to program this feature with a loudspeaker DN, but operation will be the same as for direct dial to a loudspeaker DN.

Feature packaging

This feature requires Voice Call requires the Optional Features (OPTF) package 1.

Feature implementation

Task summary list

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

1. Table 131: LD 11 on page 392.

   - Configure Voice Call for the originating Meridian 1 proprietary telephone.
2. Table 132: LD 15 on page 392

   Configure Handsfree Voice Call for the system.

Table 131: 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.
KEY	xx SCR yyy...y	Adds a single appearance single call key on the terminating telephone, where: xx = key number, and yyy...y = the DN assigned to the Voice Call key for the originating telephone.
KEY	xx VCC yyy...y	Adds a Voice Call key on the originating telephone, where: xx = key number, and yyy...y = the DN of the terminating telephone. This key activates the feature.

Table 132: LD 15

Prompt	Response	Description
REQ:	CHG	Change.

Prompt	Response	Description
TYPE:	FTR	Features and options
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- OPT	(HVD) HVA	Handsfree Voice Call (denied) allowed.

Feature operation

Voice Call

To make a Voice Call:

- Lift the handset and press Voice Call. The DN is automatically dialed. If the called telephone is busy on another DN, you hear continuous ringing. If the telephone is busy on the Voice Call DN, you hear busy tone.

To end a Voice Call:

- Press Rls.

To answer a Voice Call on an idle telephone:

- Let the call ring once. The call is answered automatically, activating the Voice Call DN over the speaker. For a two-way conversation, lift the handset.

  If busy on another DN, you hear a single beep and the Voice Call DN flashes. You must end your present call to receive the Voice Call.

Handsfree Voice Call

HVA option

The originating telephone (telephone A) places a VCC/DIG call to the destination telephone (telephone B).

1. Telephone B rings once.
2. After one ring, telephone B automatically answers the call in Handsfree mode.

The DN and Handsfree LCDs are lit and a two-way voice path is established.

HVD option

Telephone A places a call to telephone B.

1. Telephone B rings once.
2. After one ring, telephone B automatically answers the call in loudspeaker only mode.

   The DN LCD is lit and the Handsfree LCD remains dark, establishing a one-way voice path from telephone A to telephone B. At this point, telephone A is unable to hear the person at telephone B.

   To reestablish a two-way voice path, telephone B must either go off-hook or press the Handsfree button.

   Busy calls are not changed by Handsfree Voice Call.

Chapter 63: VTRK Failover Upon Network Failure

Contents

This section contains information on the following topics:

Feature description

The VTRK Failover Upon Network Failure feature enables Virtual Trunk (VTRK) calls to survive a network failure.

Upon network failure, the Leader Signaling Server is isolated from the rest of the network, which caused it to unregister its VTRKs from the Call Server. In the meantime, the Follower Signaling Server did not receive the Leader I'm Master broadcast message, so an election is called and the Follower Signaling Server becomes the master. As a result, the VTRK application attempts to register with the Call Server. The Call Server grants the request since the Leader Signaling Server already sent a server offline message. VTRK calls are now made through the Follower Signaling Server.

Once the network connection is reestablished, an election is called again. The Follower Signaling Server relinquishes mastership and unregisters its VTRKs from the Call Server. The Leader Signaling Server is able to reregister VTRKs. VTRK calls are now made through the Leader Signaling Server again.

The VTRK Failover Upon Network Failure feature implements the VTRK Network Health Monitor to monitor the health of the network, which is configured using Element Manager.

The VTRK Network Health Monitor task starts when the Signaling Server boots. The VTRK Network Health Monitor task receives a request from the Signaling Server applications to monitor the connectivity status of a given IP address. The Virtual Trunk uses this monitor task to send ping messages to the list of preconfigured IP addresses.

The monitored IP addresses are read from the config.ini file. The following example shows a sample of the new section in the config.ini file with VTRK Network Health Monitor task configured using Element Manager.

Config.ini file with VTRK Network Health Monitor task enabled

```
[VTRK NETMON]

ENABLE = 1

IP = 192.168.2.1

IP = 192.168.2.10
```

If VTRKNETMON section is not present, or if there is no valid IP address, then the TLAN Gateway is monitored by default.

The following parameters apply to the config.ini file:

- ENABLE—specifies if the VTRK Network Health Monitor monitors IP addresses or not. If there is no value present, the IP addresses are monitored by default.
    - ENABLE = 1 (VTRK Network Health Monitor monitors IP addresses)
    - ENABLE = 0 (VTRK Network Health Monitor does not monitor IP addresses)
- IP — specifies all IP addresses to be monitored by the VTRK Network Health Monitor. Invalid IP address formats are ignored.

If at least one IP address in the monitored list is reachable and the Signaling Server is Leader, the Virtual Trunks remain registered with the Call Server. If all monitored IP addresses are unreachable, the Leader Signaling Server unregisters its Virtual Trunks to enable the Follower Signaling Server to register with the Call Server and take over the Virtual Trunk operation.

The VTRK Network Health Monitor sends ICMP PING requests every ten seconds to each monitored IP address. If no response is received within five seconds, five more PING retries are sent, one second apart. If there is still no response, the IP address is considered unreachable. The detection time for each IP address is between 10 and 20 seconds.

When all monitored IP addresses are determined unreachable and the Signaling Server is Leader, the Virtual Trunks are unregistered from the Call Server. In the meantime, this isolated Signaling Server continues to monitor the network status by pinging all monitored IP addresses every ten seconds. When the network recovers, the Signaling Server reregisters the Virtual Trunks if it is the node Master. If only a subset of all monitored IP addresses are unreachable, the Signaling Server maintains its Virtual Trunk registration, and all IP addresses, whether they are reachable or not, are pinged every ten seconds.

As soon as one ping response is received, the corresponding IP addresses is determined reachable. If the isolated Signaling Server is the Leader, it attempts to register its Virtual Trunks with the Call Server. If this Signaling Server is the Follower, it ignores this event and continues to monitor the list of IP addresses by sending a PING request every ten seconds.

Although the monitored IP addresses are pinged separately, they are read from the config.ini file and are monitored at the same time when the Signaling Server boots. Thus, the time to detect a network failure is between 10 and 20 seconds

Administration and maintenance

The oam CLI prompt is added to the oam > shell.

Table 133: New CLI prompt

Prompt	Response	Description
`oam`	`vtrkNetMonShow`	Print the current list of monitored IP addresses and their status.

Element Manager configuration

Use Element Manager to configure the following new functionalities:

- VTRK Network Health Monitor configuration for an IP Telephony node
- VTRK Network Health Monitor CLI commands support for an IP Telephony node

VTRK Network Health Monitor configuration for an IP Telephony node

Log on to Element Manager. Navigate to **IP Network > Nodes: Servers, Media Cards** to open the Node Configuration window to begin configuration.

Adding an IP address to VTRK Network Health Monitor

1. Enter a node number in the **New Node** field or expand a configured node.
2. Expand the VTRK Network Monitor configuration window.
3. Click **Add**.
4. Expand the Virtual Trunk Network Health Monitor configuration window.

   The **Monitor** check box is selected by default.
5. Click **Add** beside Monitored IP address to add a new IP address field.

   Once the maximum of 8 IP address fields is reached, the **Add** button is disabled.

6. Enter a valid IP address.
7. Click **Save and Transfer**.

Editing a monitored IP address in VTRK Network Health Monitor

1. In the Node Configuration window, click the **Edit** button of a node.
2. Expand the VTRK Network Monitor configuration window.
3. Edit the configured monitored IP address.
4. Select or deselect the **Monitor** check box.

   If the Monitor check box is deselected, the **Add** button beside Monitored IP address and the **Add** button beside each monitored IP address rows is disabled.
5. Click **Save and Transfer**.

Viewing new CLI commands in Element Manager

1. Log on to Element Manager. Navigate to **IP Network > Maintenance and Reports**.
2. Expand a configured node.
3. Click **GEN CMD** against a Signaling Server element.
4. Select **Vtrk** from the Group dropdown list.
5. Select **vtrkNetMonShow** from the Command drop down list.
6. Click **Run**.

Deleting a monitored IP address in VTRK Network Health Monitor

1. In the Node Configuration window, expand the Node.
2. Expand the VTRK Network Monitor configuration window.
3. Click the **Remove** button beside a configured monitored IP address to delete it.
4. Click **Save and Transfer**.

Table 134: VTRK Network Monitor button description in Element Manager

Action	Response	Comment
Monitor	Check box	Selected by default. If selected, ENABLE = 1 If deselected, ENABLE = 0.
IP address	128 bit IPv4 address in the format: xxx.xxx.xxx.xxx	Each row caption is suffixed with the number of rows. Up to 8 IP addresses are configured. Config.ini stores these values in the following format: [VTRK NETMON] ENABLE = 1 IP =

Action	Response	Comment
		47.11.221.22 IP = 47.11.221.23
Click the Check box	If the check box is selected, the Add and Remove buttons are enabled. If the check box is deselected, the Add and Remove buttons are disabled.	When the Monitored check box is deseleted, all configured IP addresses, and the Add and Remove buttons are disabled.
Click the Add button	A new row appears to enter a Monitored IP address.	The Add button is enabled if the number if IP address rows are less than 8. The Add button is disabled after 8 rows of IP addresses are displayed and when the Monitor check box is deselected.
Click the Remove button	The Monitored IP address row is deleted and the next row caption changes to reflect the current row number.	Click the Remove button beside the Monitored IP address to delete the IP address. If the Monitor check box is deselected, the Remove button is disabled.

VTRK Network Health Monitor Configuration CLI commands

Log on to Element Manager. Navigate to **IP Network > Maintenance and Reports**.

Running the vtrkNetMonShow command

1. Expand a configured node.
2. Click the **GEN CMD** button against a Signaling Server element.
3. Select **vtrk** from the Group drop down list.
4. Select **vtrkNetMonShow**.
5. Click **Run**.

New SNMP alarms

The following new SNMP alarms are created on the Signaling Server:

- ITG4121

  Message: “Monitored IP: <%s>: Fail Retry” Description: this is created when a ping was sent to an IP address and no response was received within five seconds. Five more pings are sent and didn't receive response in Five seconds. Five more pings are to be sent one second apart before it is determined that the IP address is unreachable. This alarm is created by tNetMon task. Severity: Warning

- ITG5121

  Message: “Monitored IP <%s>: OK” Description: this is created when a ping response was received for a previously unreachable IP address. This alarm is reported by tNetMon Task Severity: Clear

- ITG1121

  Message: “Monitored IP <%s>: Unreachable” Description: this is created when five ping retries failed; that is, no response was received. This alarm is reported by tNetMon Task Severity: Critical

- ITG1122

  Message: “Virtual Trunk Network is: Down” Description: this alarm is created by VTRK task when all the monitored IP addresses are determined to be unreachable and this Signalling Server is current node master.Severity: Critical

- ITG5122

  Message: “Virtual Trunk Network is: OK” Description: this alarm is created by VTRK task when the first IP reachable event occurs to clear the isolation alarm. Severity: Clear

Chapter 64: X08 to X11 Gateway

Contents

This section contains information on the following topics:

Feature description

X08/X11 Gateway is a feature which allows the use of both Generic X08 and Generic X11 software in the same network. This feature allows individual system nodes, running X08 and X11 software, to interface with one another. The Gateway makes this interconnection possible by allowing X11 nodes to bridge between both R2 Multifrequency Compelled (R2/MFC) signaling and L1 signaling, and Integrated Services Digital Network (ISDN) signaling. Although certain configurations of the X08 nodes may be necessary, no changes to X08 software are required.

This feature provides connectivity between X08 and X11 nodes, using L1, R2 Multifrequency Compelled (MFC), and Integrated Services Digital Network (ISDN) signaling protocols. The X08 L1 Signaling supports call setup, a numbering plan and Calling Number Identification (CNI). However, the L1 Signaling that is provided into X11 is a subset of the X08 L1 Signaling, supporting only the supplementary services required to support CNI and the suppression of Bring Up Receiver (BUR) signals.

Figure 14: R2 MFC Connections and Tandems on page 402 summarizes the types of R2 MFC connections and tandems that are supported by the X08 to X11 Gateway.

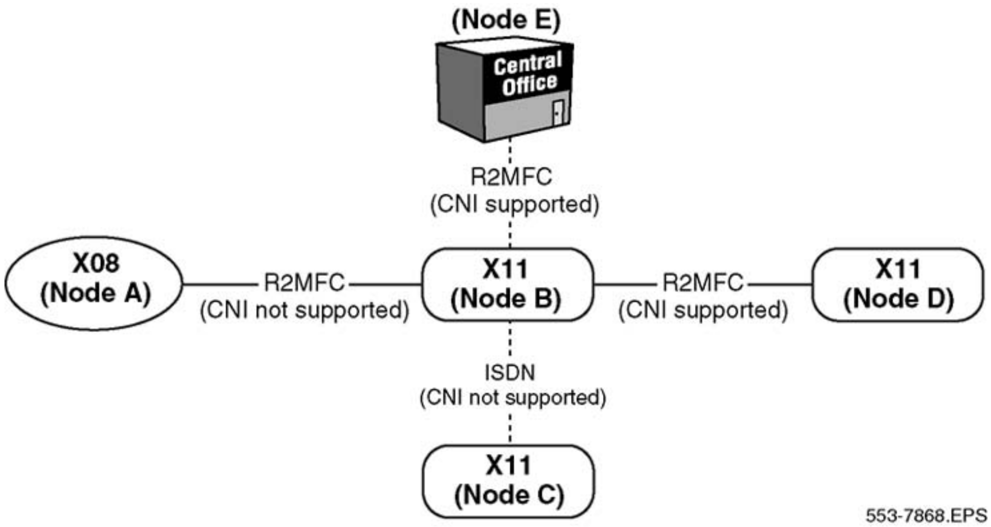


Figure 14: R2 MFC Connections and Tandems

X08 to X11 connections using R2 MFC routes (Node A to Node B) – CNI is not supported because X08 does not provide outgoing CNI signaling.

X11 to a private exchange using R2 MFC routes (Node B to Node E) – CNI is supported in both directions (DID/DOD).

Tandems using R2 MFC and ISDN routes, as follows:

- Tandems from an X08 node to an X11 node using R2 MFC routes to another X11 node using ISDN routes (Node A to Node B to Node C). CNI is not supported for this tandem.
- Tandems from one X11 node to another X11 node using R2 MFC routes to another X11 node using ISDN routes (Node D to Node B to Node C). CNI is not supported for this tandem.
- Tandems from a Public Exchange to an X11 node using R2 MFC CO routes to another X11 node using ISDN routes (Node E to Node B to Node C).

Figure 15: L1 MFC Connections and Tandems on page 403 summarizes the types of L1 connections and tandems that are supported by the X08 to X11 Gateway (tandemning to X11 nodes using the R2 MFC Signaling is not allowed):

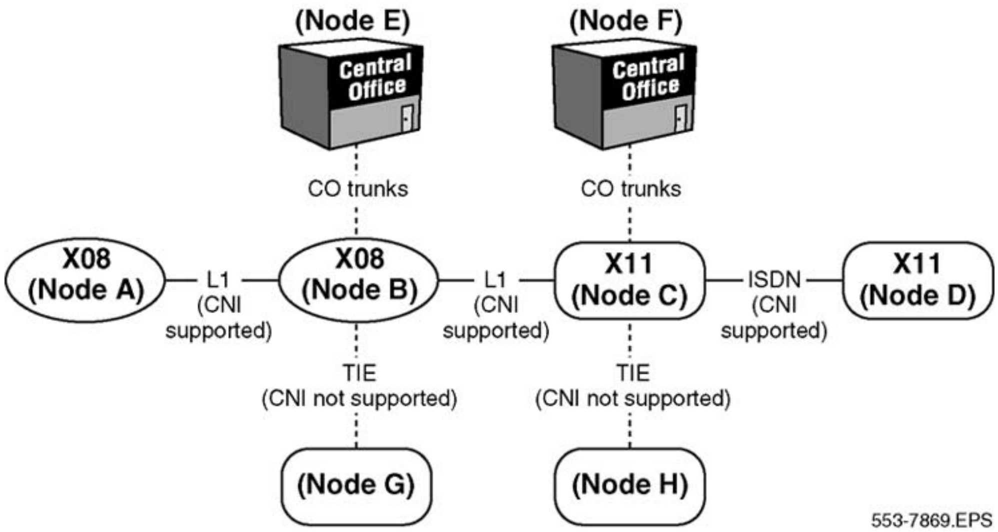


Figure 15: L1 MFC Connections and Tandems

X08 to X11 connections using L1 routes (Node B to Node C) – CNI is supported.

X08 or X11 connections to a private exchange using CO routes (Node B to Node E or Node C to Node F) – These routes can be analog or digital, and are non-R2MFC. CNI is not supported.

X08 or X11 connections to a private exchange using TIE routes (Node B to Node G or Node C to Node H) – These routes can be analog or digital, and are non-R2MFC. CNI is not supported.

Tandems using L1 and ISDN routes, as follows:

- Tandems from an X08 node to an X11 node using L1 routes to another X11 node using ISDN routes (Node B to Node C to Node D). CNI is supported.
- Tandems from an X08 node to an X11 node using L1 routes to a private exchange using analog or digital routes (Node B to Node C to Node F). CNI is not supported for this tandem.
- Tandems between X08 and X11 nodes using L1 routes to a node using TIE routes (Node B to Node C to Node H, and Node C to Node B to Node G). CNI is not supported.

Operating parameters

Routes using R2/MFC signaling can only be tandemed to routes using L1 signaling in cases where:

- the L1 route uses L1 Basic signaling (no supplementary services);
- the X11 node makes no Calling Number Identification (CNI) requests;

- the X08 node makes no call extensions for Ring Again (RGA); and
- signal assignment is co-ordinated between the X11 and X08 nodes.

L1 signaling in X11 must use TIE trunks.

L1-signaled routes will support CNI only when End-to-End Signaling is used.

The following groups of features do not operate on L1-signaled calls between X08 and X11 nodes:

- features requiring Bring Up Receiver (BUR) signals;
- call diversions;
- X08 trunk optimization;
- call transfer to an unestablished connection;
- Break-in, Recall, Incoming Call Identification (ICI) requests and Night Service Notification attendant features; and
- Ring Again (RGA).

X08 L1 signaling allows only one unsupervised trunk in a call connection. An X11 node tandeming an L1 connection from an X08 node does not inform the X08 node of unsupervised-trunk usage.

R2/MFC tandems support End-to-End Signaling only when the tandem node uses either the same R2/MFC table for both trunks or uses two tables with identical contents and the same End-to-End Signaling code. Calling Number Identification (CNI) is carried end-to -end even where End-to-End Signaling is not available.

X08 to X11 connections, using R2/MFC, do not support CNI. Outgoing CNI, on a tandem R2/MFC connection from an X08 node, uses the customer identifier of the tandeming X11 node, plus the Access Code of the route from the X08 node.

CNI is not supported over R2/MFC to ISDN tandem connections.

A third level of R2/MFC signaling, consisting only of backward signals, is not supported. This level of signaling is used for coin-box calls or calls from subscribers with home meters.

CNI in Call Detail Recording (CDR) records will have the same length only when all DNs, route access codes, trunk identifiers and attendant identifiers have the same length.

X08 does not have Integrated Services Digital Network (ISDN) capabilities.

Feature interactions

The network supported features using Gateway depends on the specific types of connection involved in any particular call.

Calling Number Identification (CNI)

Calling Number Identification (CNI) is supported on R2/MFC signaling connections between X11 nodes and Central Offices (COs), in both directions of calling, provided that the trunk being used has CNI-allowed Class of Service. CNI has the following characteristics across this type of connection:

- CNI begins with an optional customer identifier, 1-8 digits long;
- the customer identifier is followed by a caller identifier (a DN of 0-7 digits, an attendant identifier, a trunk identifier or a route access code);
- the attendant identifier has a maximum of 4 digits (identified on a customer basis); if the attendant identifier has not been defined, the attendant DN is used;
- the trunk identified has 0-7 digits (as assigned in LD 14);
- the trunk identifier does not have a unique value;
- the route access code is used if the trunk identifier has not been defined;
- a maximum of 16 digits of CNI can be carried across an R2/MFC connection, and
- end-to-end CNI to the CO works when the call tandems across more than one X11 node, using R2/MFC.

CNI is not supported on tandem connections between R2/MFC and ISDN routes.

In R2/MFC connections between X08 and X11 nodes, end-to-end CNI is only supported in cases where it has been requested by the X08 node. The X08 node will not support outgoing CNI. On an outgoing connection, the CNI supplied to the far end is that of the tandeming node when the tandeming node has not received CNI on an R2/MFC connection.

CNI is fully supported on R2/MFC connections between X11 nodes. End-to-end CNI to a CO works when the call tandems across more than one X11 node, using R2/MFC. If a tandeming node does not receive CNI, that node sends its own CNI forward.

CNI is supported on L1-signaled routes only when End-to-End Signaling is used.

Network Ring Again

Network Ring Again is not supported across any R2/MFC signaling connection or across L1-signaled connections between X08 and X11 nodes.

Feature packaging

The following packages are required for X08/X11 Gateway:

For R2/MFC Signaling, the following package is required:

- R2/MFC package 128

For L1 Signaling, the following packages are required:

- L1 package 188 and
- R2/MFC package 128.

For R2/MFC—ISDN Gateway, the following packages are required:

- Integrated Services Digital Network (ISDN) package 145;
- Primary Rate Access (PRA) package 146;
- DID-to-Network package 161; and
- R2/MFC package 128.

For L1—ISDN Gateway, the following packages are required:

- R2/MFC package 128;
- L1 package 188;
- Integrated Services Digital Network (ISDN) package 145;
- Primary Rate Access (PRA) package 146;
- DID-to-Network package 161; and
- Network Attendant Service (NAS) package 159 is required when the L1—ISDN Gateway must transport CNI.

Feature implementation

Task summary list

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

1. Table 135: LD 14 on page 407

Assign trunk for X08 to X11 gateway.

2. Table 136: LD 16 on page 407

Assign trunk route for X08 to X11 gateway.

Table 135: LD 14

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	TIE	TIE 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
TKID	nnnnnnn	Trunk Type Identifier (Does not have to be unique).

Table 136: LD 16

Prompt	Response	Description
REQ	NEW CHG	Add, or change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large Systems.
TKTP	TIE	TIE trunk type.
CCNI	(NO) YES	Call Number Indicator or CNI enabled on route.

Feature operation

No specific operating procedures are required to use this feature.

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