Features and Services Fundamentals — Book 1 of 6 (A to B)
Avaya Communication Server 1000

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Contents

Chapter 1: New in this release

The following sections describe what's new in *Avaya Features and Services Fundamentals - Book 1 of 6, NN43001-106* for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.6.

Features

There have been no updates to the feature descriptions in this document.

Other changes

The following features have been renamed and moved from *Avaya Features and Services Fundamentals - Book 4 of 6, NN43001-106* to this technical publication:

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

Revision History

May 2013	Standard 06.03. This document is up-issued to include updates to the list of available Backup Rule types.
March 2013	Standard 06.02. This document is up-issued to support Communication Server 1000 Release 7.6.

March 2013	Standard 06.01. This document is up-issued to support Communication Server 1000 Release 7.6.
December 2011	Standard 05.04. This document is up-issued to support the removal of End of Life (EoL) and Manufactured Discontinued (MD) hardware content and associated diagrams.
May 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 This document is up-issued to remove legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.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.
May 2009	Standard 03.02 This document is up-issued to support Communication Server 1000 Release 6.0.
August 2008	Standard 02.05. This document is up-issued to add information in chapter Call Forward to Trunk Restriction.
February 2008	Draft 02.04. This document has been up-issued to support Communication Server Release 5.5.
December 2007	Standard 02.03. This document has been up-issued to support Communication Server Release 5.5.
July 2007	Standard 01.04. This document is up-issued (revising the 500 Telephone Features and Bandwidth Management Support for Network Wide Virtual Office chapters in Book 1) and revising the Conference Warning Tone Enhancement chapter in Book 2.
June 2007	Standard 01.03. This document is up-issued (revising the Software Licenses chapter in Book 6).
June 2007	Standard 01.02. This document is up-issued (revising the Network Music feature implementation in Book 5).
May 2007	Standard 01.01. This document is 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 Fundamentals - Book 1 of 6 (NN43001-106-B1). • Features and Services Fundamentals - Book 2 of 6 (NN43001-106-B2). • Features and Services Fundamentals - Book 3 of 6 (NN43001-106-B3). 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: • 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 up-issued to support 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 1 of a 3 book set.
January 2002	Standard 10.00. This document 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
1.5 Mbit Digital Trunk Interface	75	PBXI	5
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• Reference Clock Switching (see also packages 129, 131, and 154)			
16-Button Digitone/Multifrequency Telephone	144	ABCD	14
• 16-Button Digitone/Multifrequency Operation			
2 Mbit Digital Trunk Interface	129	DTI2	10
• DID Recall features on DTI2 for Italy – DID Offering			
• DID Recall features on DTI2 for Italy – DID Recall			
• Italian Central Office Special Services (see also packages 131, and 157)			
• Italian Periodic Pulse Metering			
• Pulsed E&M DTI2 Signaling			
• Reference Clock Switching (see also packages 75, 131, and 154)			
• R2MFC 1.5 Mbps DTI			
• 2 Mbps Digital Trunk Interface			
• 2 Mbps Digital Trunk Interface Enhancements:			
- Alarm Handling on DID Channels			
- Alarm Handling on Incoming COT/DID Calls			
- Call Clearance			
- Clock Synchronization			
- DID Call Offering			
- Disable Out-of-Service Alarm State			
- Fault Signal			
- Incoming Seizure			
- Outpulsing Delay			
- Release Control			

Package Name	Number	Mnemonic	Release
- Signal Recognition			
- Trunk Entering Alarm Status/Trunk Pack Exiting Alarm Status			
- 64 Kbps Alarm Indication Signal (AIS) Handling			
2.0 Mbit/s Primary Rate Interface	154	PRI2	14
• Reference Clock Switching (see also packages 75, 129, and 131)			
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• Call Hold, Permanent			
• 2500 Set Features			
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• 500 Set Features			
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• AC15 Recall: Transfer from Norstar			
• AC15 Recall: Transfer from Meridian 1			
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Advanced ISDN Network Services	148	NTWK	13
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• Advice of Charge Real-time Supplementary Services for NUMERIS and SWISSNET (see also package 101)			
• Alternative Conference PAD Levels			
• Alternative Loss Plan			
• Alternative Loss Plan for China			
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• CLID on Analog Trunks for Hong Kong (A-CLID)			
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 Alternative Answering			
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Attendant Announcement	384	AANN	25.40
• Attendant Announcement			
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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)			
• Network Individual Do Not Disturb (see also packages 9, and 159			
• Attendant Busy Verify			
• Attendant Call Selection			
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• Attendant Consoles			
• Attendant Delay on Hold			
• Attendant Display of Speed Dial or Autodial			
Attendant Forward No Answer	134	AFNA	14
• Attendant Forward No Answer			
• Attendant Forward No Answer Expansion			
• Attendant Incoming Call Indicators			
• Attendant Interpositional Transfer			
• Attendant Lockout			
Attendant Overflow Position	56	AOP	1

Package Name	Number	Mnemonic	Release
• Attendant Overflow Position			
• Attendant Position Busy			
• Attendant Recall			
• Attendant Recall with Splitting			
Attendant Remote Call Forward	253	ARFW	20
• Call Forward, Remote (Network and Attendant Wide)			
• Attendant Secrecy			
• Attendant Splitting			
• Attendant Trunk Group Busy Indication			
• Audible Reminder of Held Calls			
Autodial Tandem Transfer	258	ATX	20
• Autodial Tandem Transfer			
Automatic Answerback	47	AAB	1
• Automatic Answerback			
• Automatic Call Distribution Answer Time in Night Service			
• Automatic Call Distribution Call Delays (see also package 40)			
• Automatic Call Distribution Call Priority (see also package 40)			
• Automatic Call Distribution Call Waiting Thresholds (see also packages 40 and 41)			
• Automatic Call Distribution Calls on Hold (see also package 40)			
• Automatic Call Distribution Dynamic Queue Threshold (see also package 40)			
Automatic Call Distribution Enhanced Overflow	178	EOVF	15
• Automatic Call Distribution Enhanced Overflow			
Automatic Call Distribution Load Management	43	LMAN	1
• Automatic Call Distribution Load Management Reports			
Automatic Call Distribution Night Call Forward without Disconnect Supervision	289	ADSP	23
• Call Processor Input/Output)			
Automatic Call Distribution Package C	42	ACDC	1
• Automatic Call Distribution Report Control (see also package 50)			
• 500/2500 Line Disconnect			

Package Name	Number	Mnemonic	Release
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Automatic Call Distribution Package D, Auxiliary Security	114	AUXS	12
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• Automatic Call Distribution Activity Code			
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• Automatic Call Distribution			
Automatic Call Distribution, Package B	41	ACDB	1
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• Automatic Call Distribution Least Call Queuing			
• Automatic Call Distribution Threshold Visual Indication (see also packages 40, and 131)			
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• Automatic Call Distribution Priority Agent			
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• ACD Timed Overflow			
• Automatic Gain Control Inhibit			
• Automatic Guard Detection			
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Automatic ID of Outward Dialing	3	AIOD	1
Automatic Installation (Option 11 only)	200	AINS	16
• Automatic Installation			
Automatic Line Selection	72	LSEL	4
• Automatic Line Selection			

Package Name	Number	Mnemonic	Release
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• Automatic Number Identification Route Selection			
Automatic Number Identification	12	ANI	1
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• Automatic Number Identification on DTI			
• Automatic Preselection of Prime Directory Number			
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• Automatic Redial			
• Automatic Timed Reminders			
Automatic Wake-Up	102	AWU	10
• Automatic Wake Up			
Auxiliary Processor Link	109	APL	10
• Auxiliary Processor Link			
• Auxiliary Signaling			
• B34 Dynamic Loss Switching (see also packages 164 and 203)			
Background Terminal	99	BGD	10
• Background Terminal Facility			
Basic Alternate Route Selection	57	BARS	1
• Network Alternate Route Selection/Basic Alternate Route Selection Enhancement – Local Termination (see also package 58)			
Basic Authorization Code	25	BAUT	1
• Basic Authorization Code			
Basic Automatic Call Distribution	40	BACD	1
• Automatic Call Distribution Alternate Call Answer			
• Automatic Call Distribution Call Delays (see also package 131)			
• Automatic Call Distribution Call Priority (see also package 131)			
• Automatic Call Distribution Call Waiting Thresholds (see also packages 41, and 131)			
• Automatic Call Distribution Calls on Hold (see also package 131)			
• Automatic Call Distribution Dynamic Queue Threshold (see also package 131)			
• Automatic Call Distribution Enhancements			

Package Name	Number	Mnemonic	Release
• Automatic Call Distribution in Night Service			
• Automatic Call Distribution Threshold Visual Indication (see also packages 41, and 131)			
• INIT Automatic Call Distribution (ACD) Queue Call Restore			
Basic Call Processing	0	BASIC	1
Basic Queuing	28	BQUE	1
• Basic Queuing			
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Basic Routing	14	BRTE	1
• Basic Routing			
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• Integrated Services Digital Network Basic Rate Interface (see also packages 216, and 233)			
• ISDN Basic Rate Interface Connected Line Presentation/ Restriction			
• Bridging			
• Busy Lamp Field Array			
Business Network Express	367	BNE	25
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• Business Network Express/EuroISDN Explicit Call Transfer			
• Business Network Express/Name and Private Number Display			
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• Busy Tone Detection for Asia Pacific and CALA			
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• Call Center Transfer Connect			

Package Name	Number	Mnemonic	Release
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• Call Detail Recording Enhancement			
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• Call Detail Recording Expansion			
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Call Detail Recording Queue Record	83	CDRQ	3
• ACD CDR Queue Record			
Call Detail Recording, Data Link	6	CLNK	1
Call Detail Recording	4	CDR	1
• Call Detail Recording			
• Call Detail Recording Enhancement			
• Call Detail Recording on Redirected Incoming Calls			
• Call Detail Recording with Optional Digit Suppression			
• Call Detail Recording 100 Hour Call			
• NPI and TON in CDR Tickets			
• Call Forward and Busy Status			
• Call Forward Busy			
• Call Forward by Call Type			
• Call Forward External Deny			
• Call Forward No Answer, Second Level			
• Call Forward No Answer/Flexible Call Forward No Answer			
• Call Forward Save on SYSLOAD			
• Call Forward Save on SYSLOAD			
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• Call Forward, Internal Calls			
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• Camp-On			
• Camp-on to Multiple Appearance Directory Number			
• Capacity Expansion			
• Card LED Status			

Package Name	Number	Mnemonic	Release
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• Centralized Attendant Services - Main			
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• Centralized Attendant Services – Remote			
• Centralized Multiple Line Emulation			
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• Charge Account and Calling Party Number			
Charge Account/Authorization Code	24	CAB	1
• Charge Account/Authorization Code Base			
• Charge Display at End of Call (see also package 101)			
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• China Number 1 Signaling – Toll Operator Break-In (see also Package 127)			
• China Number 1 Signaling Enhancements			
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Package Name	Number	Mnemonic	Release
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• Core Network Module			
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• Call Hold, Deluxe			
• Call Hold, Individual Hold Enhancement			
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Package Name	Number	Mnemonic	Release
Dial Intercom	21	DI	1
• Dial Intercom			
• Distinctive Ringing for Dial Intercom			
• Dial Pulse/Dual-tone Multifrequency Conversion			
Dial Tone Detector	138	DTD	10
• Dial Tone Detection			
• Flexible Dial Tone Detection			
Dialed Number Identification System	98	DNIS	10
• Dialed Number Identification Services			
• Dialed Number Identification Services Length Flexibility			
• Dialed Number Identification Services Name Display (see also packages 95, and 131)			
• 7 Digit DNIS for MAX			
• N Digit DNIS			24
Digit Display	19	DDSP	1
• Digit Display			
Digital Access Signaling System 2	124	DASS2	16
• Analog Private Network Signaling System (APNSS) (see also packages 190, 122, and 123)			
• DASS2/DPNSS1 – Integrated Digital Access (see also packages 122, and 123)			
Digital Private Network Signaling Network Services (DPNSS1)	231	DNWK	16
• Attendant Call Offer			
• Attendant Timed Reminder Recall and Attendant Third Party Service			
• Call Back when Free and Next Used			
• D-channel Handler Interface Expansion			
• Extension Three-Party Service			
• Loop Avoidance			
• Redirection			
• Route Optimization			
• Step Back on Congestion			
• Diversion			

Package Name	Number	Mnemonic	Release
• Night Service			
• Route Optimisation/MCDN Trunk Anti-Tromboning Interworking			
Digital Private Network Signaling System 1 Message Waiting Indication	325	DMWI	23
• DPNSS1 Message Waiting Indication			
Digital Private Network Signaling System 1	123	DPNSS	16
• Analog Private Network Signaling System (APNSS) (see also packages 190, 122, and 124)			
• DASS2/DPNSS1 – Integrated Digital Access (see also packages 122, and 124)			
• Digital Trunk Interface Enhancements			
• Digitone Receiver Enhancements: – Digitone Receiver Time-out Enhancement			
• Digitone Receiver Enhancements: – Quad Density Digitone Receiver Card			
Direct Inward Dialing to TIE (Japan only)	176	DTOT	16
• Direct Inward Dialing to TIE			
• Direct Inward Dialing to TIE Connection			
Direct Inward System Access	22	DISA	1
• Call Park on Unsupervised Trunks			
• Direct Inward System Access			
• Direct Inward System Access on Unsupervised Trunks			
Direct Private Network Access	250	DPNA	21
• Direct Private Network Access			
Directed Call Pickup	115	DCP	12
• Call Pickup, Directed			
• Directory Number Delayed Ringing			
Directory Number Expansion (7 Digit)	150	DNXP	13
• Directory Number Expansion			
• Directory Number			
- Flexible Attendant Directory Number			
- Listed Directory Numbers			
- Single Appearance Directory Number			

Package Name	Number	Mnemonic	Release
- Multiple Appearance Directory Number			
- Prime Directory Number			
• Diskette Overflow Warning			
• Display of Calling Party Denied			
Distinctive Ringing	74	DRNG	4/9
• Distinctive/New Distinctive Ringing			
Do Not Disturb, Group	16	DNDG	1
• Do Not Disturb Group			
Do Not Disturb, Individual	9	DNDI	1
• Do Not Disturb			
• Network Individual Do Not Disturb (see also packages 127, and 159)			
• Electronic Brand lining			
Emergency Services Access Calling Number Mapping	331	ESA_CLMP	23
• Emergency Services Access (See also packages 329 and 330)			
Emergency Services Access Supplementary	330	ESA_SUPP	23
• Emergency Services Access (See also packages 329 and 331)			
Emergency Services Access	329	ESA	23
• Emergency Services Access (See also packages 330 and 331)			
• End of Selection			
• End of Selection Busy			
• End-of-Dialing on Direct Inward/Outward Dialing Incoming Call Indicator Enhancement			
End-To-End Signaling	10	EES	1
• Attendant End-to-End Signaling			
• End-to-End Signaling			
Enhanced ACD Routing	214	EAR	17
• Enhanced Automatic Call Distribution Routing			
• MFC Interworking with AML Based Applications (see also packages 128, and 215)			
Enhanced Call Trace	215	ECT	18

Package Name	Number	Mnemonic	Release
• Customer Controlled Routing			
• MFC Interworking with AML Based Applications (see also packages 128, and 214)			
Enhanced Controlled Class of Service	173	ECCS	15
Enhanced DPNSS Services	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			
Trunk Verification from Station	110	TVS	9.32
• Trunk Verification from a Station			
• Uninterrupted Line Connection			
United Kingdom	190	UK	16
• Analog Private Network Signaling System (APNSS) (see also packages 122, 123, and 124)			
• UK Analogue Hardware Support			
Universal ISDN Gateways	283	UIGW	20
• Universal ISDN Gateway			
• Variable Guard Timing			
VIP Auto Wake Up	212	VAWU	17
• Hospitality Enhancements: V.I.P. Auto Wake Up			
Virtual Network Services	183	VNS	16
• Virtual Network Services			
• Virtual Network Services/Virtual Directory Number Expansion (see also package 58)			
• Voice Call			
Virtual Office	382	VIRTUAL_ OFFICE	25
• Branch Office			
• Emergency Services For Virtual Office			

Package Name	Number	Mnemonic	Release
• Internet Telephone Virtual Office			
• Virtual Office			
Virtual Office Enhancement	387	VOE	3.0
• Branch Office			
• Emergency Services For Virtual Office			
• Internet Telephone Virtual Office			
X08 to X11 Gateway	188	L1MF	15
• X08 to X11 Gateway			
Zone Call Admission Control	407	ZCAC	4.5
• Adaptive Network Bandwidth Management			

Chapter 4: 10/20 Digit ANI on 911 Calls

Contents

This section contains information on the following topics:

Feature description

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

10 digit ANI feature

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

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

20 digit ANI feature

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

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

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

II digit definition

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

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

Note:

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

CSN wireline calls format

The CSN wireline call format is as follows:

KP II NPA NXX YYYY STP

Where:

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

CSN wireless calls format

The CSN wireless call format is as follows:

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

Where:

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

Digit Display

Wireline

Wireline M911 calls display on a digital telephone as follows:

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

Wireless

Wireless M911 calls display on a digital telephone as follows:

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

- (PANI) NPA NXXX YYYY WIRELESS
- (CSN) NPA NXX YYYY*

911E (end-office) call processing

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

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

911T (tandem) call processing

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

Operating parameters

This feature is compatible with the system.

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

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

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

Feature interactions

Call Trace

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

Call Detail Recording

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

Display on CLASS telephones

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

Display on tandem call

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

Malicious Call Trace

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

Feature packaging

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

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

- Digit Display (DDSP) package 19
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Package B (ACDB) package 41
- Automatic Call Distribution Package A (ACDA) package 45
- Enhanced Automatic Call Distribution Routing (EAR) package 214
- Meridian 911 (M911) package 224
- Call Waiting Notification (CWNT) package 225
- M911 Enhancement Display (M911 ENH) package 249

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

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

  Note:

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

Note:

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

Feature implementation

Task summary list

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

1. Table 1: LD 13 - Configure MFR unit for tone detection. on page 85
2. Table 2: LD 15 - Set the Pseudo Automatic Number Identification (PANI) prompt to YES to display PANI. on page 85
3. Table 3: LD 16 - Configure the M911 ANI format. on page 85

Table 1: LD 13 - Configure MFR unit for tone detection.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	MFR	MFR unit
TN	l s c u	Format for Large System, CS 1000E system and Media Gateway 1000B, where l = loop, s = shelf, c = card, u = unit. **Note:** • MFR unit can only be configured on IPMG card 15 with unit 0 or 1. • Changes take effect only after MGC reboot.

Table 2: LD 15 - Set the Pseudo Automatic Number Identification (PANI) prompt to YES to display PANI.

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

Table 3: LD 16 - Configure the M911 ANI format.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route data block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
DES	x...x	Designator field for trunk (0-16 character alphanumeric).

Prompt	Response	Description
TKTP	TIE	Trunk type.
M911_ANI	YES	Receive ANI digits for M911 route.
M911_TRK_TYPE		Meridian 911 ANI trunk type.
	(911T) 911E	E911 tandem connections (default). End office connections.
M911_FORM		M911 ANI format.
	2 (1)	II (2 digits) +10/20-digit ANI. NPD (1 digit) +7-digit ANI (default).
....	....	

Feature operation

No specific operating procedures are required to use this feature.

Chapter 5: 16-Button Digitone/ Multifrequency Operation

Contents

This section contains information on the following topics:

Feature description

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

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

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

- authorization code
- automatic set relocation

- automatic wake-up activate
- automatic wake-up deactivate
- automatic wake-up verify
- Call Detail Recording charge account
- call forward all calls activate
- call forward all calls deactivate
- call forward all calls verify
- call forward toggle
- call park access
- conference diagnostics
- deactivate RGA, LND, SNR, or CFW
- electronic lock phone
- electronic lock phone (remote)
- Group Hunting pilot DN
- Incoming Call Identification (ICI) activate
- ICI deactivate
- ICI print
- integrated message system access
- last number redial
- maintenance access
- pick up DN
- pick up group
- pick up ringing number
- radio paging initiate (parallel)
- radio paging initiate (serial)
- radio paging answer (parallel)
- ring again deactivate
- ring again verify
- room status
- speed call controller
- speed call erase
- speed call user

- store number (erase)
- store number (redial)
- store number (save)
- system speed call user
- trunk answer from any station
- terminal diagnostics
- trunk verification, and
- user status.

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

- Call Detail Recording charge account
- call park
- Conference six trunk disconnect
- ICI override
- last number redial
- Malicious Call Trace
- override
- permanent hold
- radio paging initiate (parallel)
- radio paging initiate (serial)
- ring again activation
- speed call user
- store number (redial)
- store number (save), and
- system speed call user.

Operating parameters

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

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

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

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

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

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

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

Feature interactions

China - Flexible Feature Codes - Busy Number Redial

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

China - Flexible Feature Codes - Customer Call Forward

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

China - Flexible Feature Codes - Outgoing Call Barring

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

Flexible Feature Codes

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

Group Hunt

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

Italian Central Office Special Services

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

Feature packaging

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

Dependency:

- Flexible Feature Codes (FFC) package 139

Feature implementation

Task summary list

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

1. Table 4: LD 17 - Modify the system hardware and software parameters to enable or disable the 16-Button Digitone/Multifrequency Operation feature. on page 91
2. Table 5: LD 18 - Create or modify data for this feature in the 16-Button DTMF Data block. on page 92

Table 4: LD 17 - Modify the system hardware and software parameters to enable or disable the 16-Button Digitone/Multifrequency Operation feature.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	PARM	System Parameters.

Prompt	Response	Description
...		
PARM	(NO) YES	(No) Change to system parameters.
- ABCD	(NO) YES	16-Button DTMF (is not) is enabled.

Table 5: LD 18 - Create or modify data for this feature in the 16-Button DTMF Data block.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	ABCD	16-Button DTMF data.

Feature operation

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

Chapter 6: 2 Mbps Digital Trunk Interface

Contents

This section contains information on the following topics:

Feature description

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Periodic Pulse Metering

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

Pulsed E and M DTI2 Signaling

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

Feature packaging

2 Mbps Digital Trunk Interface (DTI2) package 129.

Feature implementation

Task summary list

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

1. Table 6: LD 14 - Create or modify trunk data blocks for DTI2 on a per trunk basis. on page 95
2. Table 7: LD 16 - Create or modify DTI2 trunk route data blocks. on page 95
3. Table 8: LD 17 - Modify the system hardware and software parameters to enable or disable the feature. on page 95
4. Table 9: LD 73 - Implement the system hardware and software parameters to enable or disable the DTI feature. on page 96

Table 6: LD 14 - Create or modify trunk data blocks for DTI2 on a per trunk basis.

Prompt	Response	Description
REQ	NEW CHG	Add. Change
TYPE	a...a	Type of data block.
SICA	(1)-16	Signaling Category table number. The category must already be defined in LD 73. Default is 16 if loop type = Japanese Digital Multiplex Interface (JDMI).
PDCA	(1)-16	Pad Category table number. The PAD category must already be defined in LD 73. Default is 16 if loop type = JDMI.
PCML	MU A	Indicate whether Mu-law or A-law Pulse Code Modulation (PCM) for voice calls is active in the channel. Not prompted for JDMI loops.

Table 7: LD 16 - Create or modify DTI2 trunk route data blocks.

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

Table 8: LD 17 - Modify the system hardware and software parameters to enable or disable the feature.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	CEQU	Common Equipment Parameters.
...		
- DTI2	0-159	2 Mbps Digital Trunk Interface (DTI) loop number.

Prompt	Response	Description
		Prompted the when DTI2 or PRI2 package is equipped.

Table 9: LD 73 - Implement the system hardware and software parameters to enable or disable the DTI feature.

Prompt	Response	Description
REQ	NEW CHG	Add. Change.
TYPE	DTI2	2 Mbps DTI.
...		

Feature operation

No specific operating procedures are required to use this feature.

Chapter 7: 2 Mbps Digital Trunk Interface Enhancements

Contents

This section contains information on the following topics:

Feature description

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

Alarm Handling on Direct Inward Dialing Channels

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

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

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

Call Clearance

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

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

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

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

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

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

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

Clock Synchronization

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

Direct Inward Dialing Call Offering

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

Disable Out-of-Service Alarm State

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

Fault Signal

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

Incoming Seizure

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

Outpulsing Delay

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

Release Control

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

Signal Recognition

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

64 Kbit Alarm Indication Signal Handling

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

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

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

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

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

Idle trunk

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

DID trunk

Trunk seized and receiving digits

The call is taken down and the trunk is idled.

Call initiated but not answered

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

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

Call answered

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

Disconnect

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

Outgoing Central Office Trunk (COT) call

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

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

Incoming COT call

Call initiated

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

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

Disconnect

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

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

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

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

Processing of overload conditions

Several enhancements occur:

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

There are no specific packaging requirements associated with this feature.

Feature implementation

Note:

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

Table 10: LD 73 - Implement the system hardware and software parameters.

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

Feature operation

No specific operating procedures are required to use this feature.

Chapter 8: 2500 Telephone Features

Contents

This section contains information on the following topics:

Feature description

This feature allows 2500 telephones (that is, basic push-button telephones without feature keys) to access features otherwise available only on Meridian 1 proprietary telephones. By dialing an octothorpe (#) and a single-digit access code, 2500 telephones can access the following features:

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

Operating parameters

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

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

Feature interactions

500 Telephone Features

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

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

Remote Call Forward

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

Feature packaging

There are no specific packaging requirements associated with this feature.

Feature implementation

Table 11: LD 10 - Enable 2500 Telephone Features.

Prompt	Response	Description
REQ:	CHG	Change.

Prompt	Response	Description
TYPE:	500	Telephone type.
TN		Terminal Number
	l s c u	Format for Large System, CS 1000E system and Media Gateway 1000B, where l = loop, s = shelf, c = card, u = unit.
CLS	(XFD) XFA	(Deny) allow transfer.
FTR	CFW xx	Call Forward All Calls and DN length (4-23). Enter X CFW to remove.
	SCC xxxx	Speed Call Controller and list number. Enter X SCC to remove.
	SCU xxxx	Speed Call User and list number. Enter X SCU to remove.
	SSU xxxx	System Speed Call User and list number. Enter X SSU to remove.
	PHD	Allow Permanent Hold. Enter X PHD to remove.

Feature operation

Call Forward All Calls

Case 1: FFC active, CFW not active

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

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

To deactivate Call Forward, follow these steps:

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

Case 2: FFC not active, CFW not active

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

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

To deactivate Call Forward, follow these steps:

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

Case 3: FFC active, CFW active

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

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

To reactivate Call Forward, follow these steps:

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

– or -

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

– or –

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

Speed Call Controller

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

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

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

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

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

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

Speed Call User

To make a Speed Call, follow these steps:

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

System Speed Call User

To make a System Speed Call, follow these steps:

1. Lift the handset and dial SPRE 73. You hear dial tone.
2. Dial the System Speed Call code (0-999).
3. The number is dialed automatically.

Permanent Hold

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

1. Flash the switchhook. You hear dial tone.
2. Dial #4.
3. Hang up.

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

Chapter 9: 500 Telephone Features

Contents

This section contains information on the following topics:

Feature description

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

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

Operating parameters

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

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

Feature interactions

2500 telephone features

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

Feature packaging

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

Feature implementation

Table 12: LD 10 - Enable 500 type telephone features.

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

Prompt	Response	Description
CLS	(XFD) XFA	(Deny) allow transfer.
FTR	CFW xx	Call Forward All Calls and DN length (4-23). Enter X CFW to remove.
	SCC xxxx	Speed Call Controller and list number. Enter X SCC to remove.
	SCU xxxx	Speed Call User and list number. Enter X SCU to remove.
	SSU xxxx	System Speed Call User and list number. Enter X SSU to remove.
	PHD	Allow Permanent Hold. Enter X PHD to remove.

Feature operation

Call Forward All Calls

To forward your calls, follow these steps:

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

To cancel forwarding, follow these steps:

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

Speed Call Controller

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

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

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

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

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

1. Lift the handset and dial SPRE + 75. You hear dial tone.
2. Dial the Speed Call code (0-999) you want to remove.
3. Hang up.

Speed Call User

To make a Speed Call, follow these steps:

1. Lift the handset and dial SPRE + 76. You hear dial tone.
2. Dial the Speed Call code (0-999).
3. The number is dialed automatically.

System Speed Call User

To make a System Speed Call, follow these steps:

1. Lift the handset and dial SPRE + 73. You hear dial tone.
2. Dial the System Speed Call code (0-999).
3. The number is dialed automatically.

Permanent Hold

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

1. Flash the switchhook. You hear dial tone.
2. Dial SPRE + 77.
3. Hang up.

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

Chapter 10: 500/2500 Line Disconnect

Contents

This section contains information on the following topics:

Feature description

500/2500 Line Disconnect

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

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

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

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

Figure 1: Incoming trunk call of internal call disconnect function when 500/2500 line disconnect feature is configured on page 116 illustrates how an incoming trunk call or internal call functions with 500/2500 Type Line Disconnect. This illustration shows the incoming trunk call or internal call disconnected and dial tone being provided by the analog (500/2500-type) port with the new Class of Service (CLS) Line Disconnect Tone Allowed (LDTA).

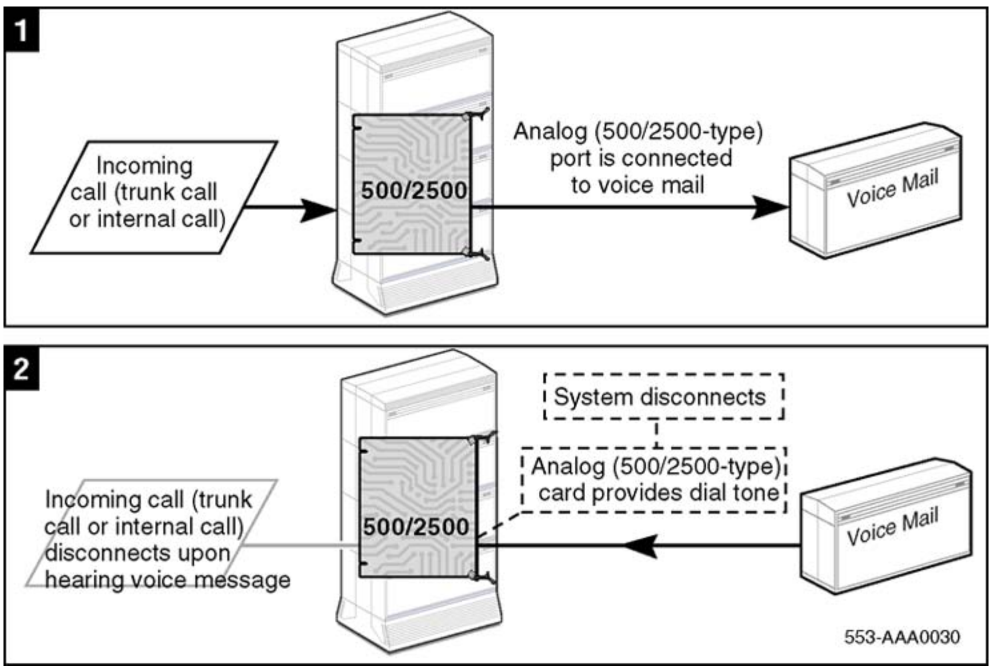


Figure 1: Incoming trunk call of internal call disconnect function when 500/2500 line disconnect feature is configured

500/2500 Line Disconnect for Outgoing Calls

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

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

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

Operating parameters

500/2500 Line Disconnect

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

500/2500 Line Disconnect for Outgoing Calls

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

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

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

Feature interactions

500/2500 Line Disconnect

500/2500 Automatic Call Distribution agent

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

Attendant Extended Call

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

Conference No Hold Conference

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

500/2500 Line Disconnect for Outgoing Calls

Attendant Extended Call

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

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

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

Call Transfer

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

Conference No Hold Conference

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

Hunting

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

Tone to Last Party

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

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

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

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 13: LD 10 - Allow Line Disconnect Tone for analog (500/2500-type) ports. on page 120
2. Table 14: LD 15 - Specify the dial tone timer for analog (500/2500-type) ports. on page 120

Note:

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

Table 13: LD 10 - Allow Line Disconnect Tone for analog (500/2500-type) ports.

Prompt	Response	Description
REQ:	NEW CHG	New. Change.
TYPE:	500	Telephone type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(LDTD) LDTA (WTA) WTD	(Deny) allow Line Disconnect Tone. (Allow) deny Warning Tone.

Table 14: LD 15 - Specify the dial tone timer for analog (500/2500-type) ports.

Prompt	Response	Description
REQ:	NEW CHG	New. Change.
TYPE:	TIM	Timers.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		

Prompt	Response	Description
- LDTT	2-(6)-30	Line Disconnect Tone timer for the 500/2500 port, in seconds.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 11: AC15 Recall: Timed Reminder Recall

Contents

This section contains information on the following topics:

Feature description

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

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

Operating parameters

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

AC15 TIE trunks must be configured on a route basis.

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

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

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

Feature interactions

AC15 Recall: Transfer from Norstar

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

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

Access Restrictions

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

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

Attendant Clearing During Night Service (ACNS)

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

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

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

Attendant Console

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

Attendant Console - Call Key Lamp State and Display

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

Attendant Forward No Answer

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

Attendant Overflow Position

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

Attendant Secrecy

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

Call Hold, Permanent

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

Called Party Name Display

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

Conference

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

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

Digit Display

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

Network Attendant Service

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

Night Service Enhancements

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

Periodic Clearing

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

Recall to Same Attendant

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

Secrecy Enhancement

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

Series Calls

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

Slow Answer Modification (SLAM)

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

Feature packaging

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

Feature implementation

Task summary list

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

1. Table 15: LD 15 - Set the Slow Answer Recall timer at the RTIM prompt. on page 128
2. Table 16: LD 16 - Define a TIE route and telephone ATRR option. on page 129
3. Table 17: LD 14 - Define an AC15 TIE trunk on an XFEM card. on page 129

Table 15: LD 15 - Set the Slow Answer Recall timer at the RTIM prompt.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	TIM	Timers Data Block.

Prompt	Response	Description
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		
- RTIM	xxx yyy zzz	xxx = timer in seconds for the Slow Answer Recall and the AC15 Timed Reminder Recall. yyy = timer in seconds for Camp-on Recall. zzz = timer in seconds for Call Waiting Recall.

Table 16: LD 16 - Define a TIE route and telephone ATRR option.

Prompt	Response	Description
REQ	NEW	New.
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	Trunk type.
DTRK	NO	Digital trunk.
TIDY	xxxx yyyy	Trunk identity.
ATRR	YES	AC15 Recall: Timed Reminder Recall. Calls transferred to an AC15 trunk on this route are subject to Timed Reminder Recall. Prompted with ACRL package 236 if TKTP = TIE and DRTK = NO.

Table 17: LD 14 - Define an AC15 TIE trunk on an XFEM card.

Prompt	Response	Description
REQ	NEW	New.
TYPE	TIE	TIE trunk.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CDEN	4D	Card density.
CUST	xx	Customer number, as defined in LD 15
SIGL	WR4	AC15 signaling.
SUPN	YES	Answer and disconnect supervision required.

Feature operation

Activate Timed Reminder Recall

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

Attendant as a controlling party

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

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

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

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

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

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

   Note:

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

Telephone as a controlling party

1. A call is established with a telephone on the system.
2. Transfer to an extension over an AC15 TIE trunk by using a flash hook on an analog (500/2500 type) telephone or pressing the TRN key on a proprietary telephone.
3. Complete the transfer before the extension answers by going on-hook on an analog (500/2500 type) telephone or pressing the TRN key on a proprietary telephone. This will start the AC15 recall timer.

Answer a Recall

Attendant

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

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

   Note:

   If the called extension answers, the recall is removed from the attendant console.

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

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

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

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

Night DN or Central Answering Position

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

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

Chapter 12: AC15 Recall: Transfer from Meridian 1

Contents

This section contains information on the following topics:

Feature description

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

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

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

Z completes the transfer and the AC15 trunk is released. Figure 2: System to Norstar call transfer on page 134 illustrates this example.

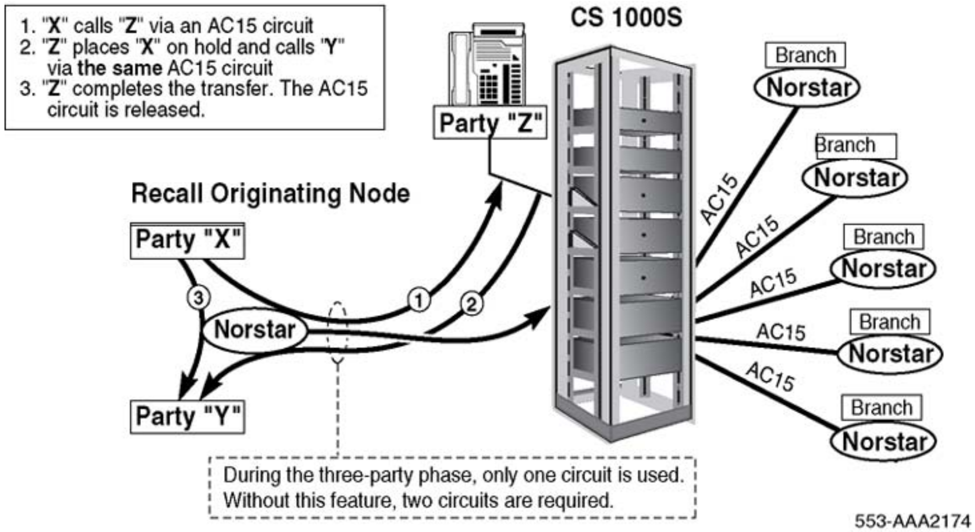


Figure 2: System to Norstar call transfer

Operating parameters

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

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

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

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

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

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

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

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

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

Optimization is not performed if a Conference key is used.

AC15 trunks using MFC signaling are not supported.

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

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

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

Feature interactions

AC15 Recall: Transfer from Norstar

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

Authorization Codes

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

Autodial Last Number Redial

Autodial and Last Number Redial are supported with the AC15 Recall: Transfer from Meridian 1 on the first transfer, provided that the digits are outpulsed on the trunk after the End-to-End

Signaling Delay timer expires. If the far end is not ready, the call will fail because no dial tone detection is performed by the system.

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

Call Park

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

Call Detail Recording

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

Conference

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

Digit Display

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

Redirection

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

Speed Call Network Speed Call

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

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

Feature packaging

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

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

Feature implementation

Task summary list

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

1. Table 18: LD 15 - Disable the End-to-End Signaling Tone to originating party at the EEST prompt. on page 138
2. Table 19: LD 16 - Define the route accepting recall signal. on page 138
3. Table 20: LD 14 - Define the AC15 trunk. on page 138
4. Table 21: LD 11 - Define the Aries telephones. on page 139

Table 18: LD 15 - Disable the End-to-End Signaling Tone to originating party at the EEST prompt.

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	FTR	Customer Features and Options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		
EEST	NO	End-to-End Signaling Tone to originating party.

Table 19: LD 16 - Define the route accepting recall signal.

Prompt	Response	Description
REQ	NEW CHG	New. 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	Trunk type requires response when REQ = NEW.
...		
CNTL	YES	Changes to controls or timers.
TIMR	EESD 0 - (1024) - 4992	End-to-End Signaling Delay timer. If EESD = 0, the timer is not started and the buffered digits will not be outpulsed.
DLTN	YES	Dial tone provided by the system to the far end switch.
TRRL	YES	Recall signal can be received and transmitted on this route.

Table 20: LD 14 - Define the AC15 trunk.

Prompt	Response	Description
REQ	NEW CHG	New. Change.
TYPE	TIE	Type of trunk.
TN		Terminal Number

Prompt	Response	Description
	l s c u	Format for Large System, Media Gateway 1000B. and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XFEM	Extended Flexible E & M trunk card.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
SIGL	WR4	AC15 signaling.
SUPN	YES	Answer and disconnect supervision required.
CLS	DTN	Digitone Class of Service.

Table 21: LD 11 - Define the Aries telephones.

Prompt	Response	Description
REQ:	NEW CHG	New. Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15
...		
KEY	0-69 TRN 0-69 NUL	To add or remove a Call Transfer key.

Feature operation

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

Chapter 13: AC15 Recall: Transfer from Norstar

Contents

This section contains information on the following topics:

Feature description

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

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

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

Operating parameters

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

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

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

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

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

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

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

Feature interactions

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

AC15 Recall: Timed Reminder Recall

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

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

AC15 Recall: Transfer from Meridian 1

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

Attendant Consoles

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

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

Break-in to Enquiry Calls

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

Call Detail Recording

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

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

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

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

Call Park

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

Call Trace

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

Call Trace Enhancement

This enhancement is applicable to the AC15 Recall: Transfer from Norstar feature. A record is issued any time the call state or the active call changes after a recall or a release message has been received from Norstar.

Calling Party Control

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

Call Transfer

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

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

Conference

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

Dial Access to Group Calls

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

Digital Private Network Signaling System 1 (DPNSS1) Route Optimization

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

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

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

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

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

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

DPNSS1 Three-party Service

When the telephone on a Norstar system completes a call transfer between two telephones located within a DPNSS1 network:

- DPNSS1 access restrictions are checked
- the telephones' displays are updated, and
- DPNSS1 route optimization after transfer can be activated.

Incoming Call Indicator Enhancement

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

Initialize

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

MFC Signaling

AC15 trunks using MFC signaling are not supported.

Music

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

Periodic Pulse Metering

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

Radio Paging

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

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

Slow Answer Recall for Transferred External Trunks

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

Feature packaging

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

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

Feature implementation

Task summary list

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

1. Table 22: LD 15 - Define the access denied intercept treatment. on page 148
2. Table 23: LD 16 - Define the route accepting recall signal. on page 148
3. Table 24: LD 14 - Define an AC15 TIE trunk. on page 148

Table 22: LD 15 - Define the access denied intercept treatment.

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

Table 23: LD 16 - Define the route accepting recall signal.

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

Table 24: LD 14 - Define an AC15 TIE trunk.

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

Prompt	Response	Description
TYPE	TIE	TIE trunk.
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XFEM	XFEM card.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
SIGL	WR4	AC15 signaling.
SUPN	YES	Answer and disconnect supervision.
CLS	DTN	Digitone Class of Service.

Feature operation

Initiate a consultation

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

Toggling during the three-party phase

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

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

Active or held party disconnects during the three-party phase

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

Complete the transfer

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

Chapter 14: Access Restrictions

Contents

This section contains information on the following topics:

Feature description

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

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

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

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

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

Class of Service restrictions

The Class of Service restrictions assigned to telephones and TIE trunks control the degree of access to and from external networks and certain features within the system. The eight possible Class of Service Access Restrictions are described in this feature module. These restrictions are applied by service change overlay programs to terminals. Table 25: Type of terminal and the corresponding overlay program for configuring Class of Service restrictions. on page 152 lists the type of terminals and the corresponding overlay program.

Table 25: Type of terminal and the corresponding overlay program for configuring Class of Service restrictions.

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

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

Fully Restricted Service

There are three levels of Fully Restricted Service:

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

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

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

Note:

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

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

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

      Note:

      The FRPT prompt in LD 17 allows or denies access to incoming calls for FRE CLS telephones. It allows FRE calls to Call Pickup, Night Answer, and to receive modified calls.
    - allowed to originate and receive internal calls
    - allowed access to TIE and CCSA networks
    - allowed access to and from the exchange network using call modification from an unrestricted telephone
- denied access (either by dialing or through an attendant) to and from the exchange network

  Note:

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

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

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

Semi-Restricted Service (SRE)

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

Toll Denied Service (TLD)

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

Conditionally Toll Denied Service (CTD)

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

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

Conditionally Unrestricted Service (CUN)

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

Unrestricted Service (UNR)

- • allowed to originate and receive calls from the exchange network

The eight possible Class of Service Access Restrictions are described in Table Table 26: Class of Service Access Restrictions chart on page 155.

Table 26: Class of Service Access Restrictions chart

	UNR	CTD/CUN	TLD	SRE	FRE	FR1	FR2
Incoming trunk calls	Yes	Yes	Yes	Yes	No Yes, if using call modification. (See Fully Restricted Service on page 152)	No	No
Outgoing non-toll trunk calls	Yes	Yes	Yes	No direct access Yes, if using attendant or UNR telephone	No direct access Yes, if using UNR telephone	No	No
Outgoing toll trunk calls (0 or 1+ on COT or FX)	Yes	No direct access Yes, if using BARS/ NARS	No direct access Yes, if using attendant or	No direct access Yes, if using attendant or UNR telephone	No direct access Yes, if using UNR telephone	No	No

	UNR	CTD/CUN	TLD	SRE	FRE	FR1	FR2
			UNR telephone				
To/From TIE trunk	Yes	Yes	Yes	Yes	Yes	Yes	No
To/From internal	Yes	Yes	Yes	Yes	Yes	Yes	Yes
BARS/ NARS calls TGAR = No	Uses NCOS only	Uses NCOS only	Uses NCOS and CLS	Uses NCOS and CLS	Uses NCOS and CLS	Uses NCOS and CLS	Uses NCOS and CLS
BARS/ NARS calls TGAR = Yes	Uses NCOS and TGAR	Uses NCOS and TGAR	Uses NCOS, CLS, and TGAR	Uses NCOS, CLS, and TGAR	Uses NCOS, CLS, and TGAR	Uses NCOS, CLS, and TGAR	Uses CLS only

Code Restriction

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

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

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

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

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

Trunk Group Access Restriction

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

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

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

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

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

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

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

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

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

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

Assume the following seven TGAR codes are required:

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

The TGAR/TARG matrix summary is as follows:

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

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

- 0 (has no restrictions)
- 1 (cannot access trunk routes 0 through 6)
- 2 (cannot access trunk routes 2 through 5)
- 3 (cannot access trunk routes 3 through 5)
- 4 (cannot access trunk routes 2 and 6)
- 5 (cannot access trunk routes 3 through 6)
- 6 (cannot access trunk routes 5 and 6)

Trunk signaling arrangements

Trunk-to-trunk connections are further controlled by the signaling and supervision arrangements assigned to each trunk. Table 27: Trunk signaling arrangements on page 159 summarizes the trunk signaling arrangements.

Table 27: Trunk signaling arrangements

From	To		
	Trunk with/ without disconnect supervision	**Paging dictation trunk**	**Telephone (non-trunk)**
Trunk with disconnect supervision	Yes	No	Yes
Trunk without disconnect supervision	No	No	Yes
RAN/Paging dictation trunk	No	No	No
Telephone	Yes	Yes	Yes

Note:

Yes = connection allowed No = connection disallowed

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

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

To transfer one outgoing external trunk to another, both external trunks must have answer and disconnect supervision, both external calls must be established and Trunk to Trunk Connection feature must be configured.

Port Access Restrictions

The port access restrictions feature prevents port-based attacks on the Avaya Communication Server 1000 (Avaya CS 1000) Release 6.0 (and later) system. Port blocking rules are automatically installed with Avaya CS 1000 Release 6.0 (and later). The port access restrictions applies to only VxWorks platforms. The port access feature completely protects the ELAN interface on the MGC and MC32S cards.

A Security Administrator can customize the port blocking rules using either LD 117 or Element Manager. There are mandatory rules that cannot be modified or deactivated. The mandatory rules are considered system essential and remain in an activated state regardless of whether the port access is configured with default or customized settings.

The port access rules can only be activated on servers with VxWorks platforms (MGC, MC32S, CP PIV and CP PM). CP PM Co-res CS and SS uses a Linux-based platform with a shell application called VxWorks (VXELL) Call Server. You cannot enable the port access restrictions rules directly for this type of server, but you can administer the port access for other VxWorks components.

There are three types of access rules that you can configure using Element Manager:

- Default: The System default configuration.
- Custom: Select this option only if a Custom settings file exists. You can create a custom file that must be an .XML based file.
- None: Selecting this rule implies that all rules are disabled (except for the mandatory system rules) and that the firewall is deactivated

For more information about configuring port access restrictions, see *Avaya Security Management Fundamentals, NN43001-604*.

Operating parameters

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

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

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

Feature interactions

AC15 Recall: Timed Reminder Recall

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

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

Call Park

A call can be parked on any DN, regardless of its Class of Service. Access to a parked call is governed by the same Class of Service restrictions for normal trunk-to-telephone call processing. Table Table 28: Parked call Access Restrictions. on page 161 details the restrictions. These restrictions can be overridden with the Authorization Code.

Table 28: Parked call Access Restrictions.

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

Call Pickup Network Wide

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

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

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

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

The Code Restriction sub-feature is not supported.

Direct Inward System Access

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

Group Hunt

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

ISDN QSIG/EuroISDN Call Completion

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

New Flexible Code Restriction

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

Scheduled Access Restrictions

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

Trunk Barring

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

Virtual Network Services

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

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

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

9. Table 37: LD 19 - Implement Code Restriction on trunk routes. on page 167
10. Table 38: LD 16 - Define toll access digits that are to be ignored for Code Restriction. on page 168

Table 29: LD 10 - Define a Class of Service and TGAR code for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	500	Telephone type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.

Table 30: LD 11 - Define a Class of Service and TGAR code for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type ? for a list of possible responses.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
CLS	(CTD) UNR CUN TLD SRE FRE FR1	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1.

Prompt	Response	Description
	FR2	Fully Restricted 2.

Table 31: LD 14 - Define a Class of Service and TGAR code for trunks.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	TIE	TIE trunk.
	ISA	Integrated Services Access trunk.
	CSA	Common Control Management Access Line.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
TGAR	0-(1)-31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.
	X	Precede with X to remove
CLS	(CTD) UNR CUN TLD SRE FRE FR1 FR2	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1. Fully Restricted 2.

Table 32: LD 88 - Assign a Class of Service to the Authorization Code classcode.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	AUB	Authcode Data Block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password (see LD 15 for description).
CLAS	0-115	Classcode number.
CLS	(CTD) UNR CUN TLD SRE FRE FR1	Conditionally Toll Denied (default). Unrestricted. Conditionally Unrestricted. Toll Denied. Semi-Restricted. Fully Restricted. Fully Restricted 1.

Prompt	Response	Description
	FR2	Fully Restricted 2.
TGAR	0-(1)-31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.
NCOS	(0)-99	Toll Restricted.

Table 33: LD 86 - Enable or disable the Trunk Group Access Restriction (TGAR) option.

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

Table 34: LD 24 - Assign a Class of Service to Direct Inward System Access (DISA) numbers.

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

Table 35: LD 17 - Allow or deny incoming calls to telephones with the FRE Class of Service for all customers.

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

Table 36: LD 16 - Add or change the TARG code for a trunk route.

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

Table 37: LD 19 - Implement Code Restriction on trunk routes.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CRB	Code Restriction Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.

Prompt	Response	Description
OABS	x x x	Outgoing digits (0-9) to be ignored.

Table 38: LD 16 - Define toll access digits that are to be ignored for Code Restriction.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
OABS	x x x	Outgoing digits (0-9) to be ignored.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 15: Activity Codes for Not Ready State

Contents

This section contains information on the following topics:

Feature description

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

Operating parameters

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

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

- Observe Agent
- Call Agent

- Answer Agent
- Answer Emergency

Note:

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

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

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

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

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

Feature interactions

Multiple Queue Assignment

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

Return to Queue on No Answer

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

Feature packaging

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

The feature packaging requirements for Meridian MAX are:

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

The feature packaging requirements for Symposium are:

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

Feature implementation

Task summary list

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

1. Table 39: LD 11- Define an Activity Code key for proprietary telephones on page 171
2. Table 40: LD 23 - Enable Activity Codes in the Not Ready State for an ACD queue. on page 172

Table 39: LD 11- Define an Activity Code key for proprietary telephones

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, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15.
...	...	
KEY	xx ACNT	xx = Key number (the ACNT key cannot be configured as 0).

Table 40: LD 23 - Enable Activity Codes in the Not Ready State for an ACD queue.

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

Feature operation

To enter an Activity Code in the Not Ready State:

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

   Note:

   The * is used to delete one digit at a time. The # symbol delete all the digits entered.
4. Press the Activity key. The activity code is sent to the system and the Activity Code key lamp goes out. This completes the activity code entry.
5. An ACD agent can enter multiple activity codes for each activity completed during any Not Ready Session.
6. Repeat steps 2 on page 172-4 on page 172 until all tasks are entered.
7. Press the Not Ready key. The Not Ready key lamp goes out and the agent is placed back into the ACD queue.

To use the Display key in the Not Ready State:

1. The agent presses the Display key. The telephone display is cleared.
2. The agent presses the Activity key. The previously entered Activity Code appears in the telephone display.
3. The agent presses the Display key twice (or presses the RLS key) to display the time and date.

Note:

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

Note:

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

Chapter 16: Alarm Management

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

Alarm Management provides the following sub features:

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

For information about the Alarm Management feature, refer to Avaya LD 117: Ethernet and Alarm Management in *AvayaSoftware Input Output – Administration* (NN43001-611).

Chapter 17: Alternative Conference Pad Levels

Contents

This section contains information on the following topics:

Feature description

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

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

Feature implementation

Table 41: LD 15 - The value of the conference pad selection must be specified.

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

Feature operation

No specific operating procedures are required to use this feature.

Chapter 18: Alternative Loss Plan for China

Contents

This section contains information on the following topics:

Feature description

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

Eight Alternative Trunk Pad Matrix Options are available to satisfy the loss plan requirements of various countries:

0 – Standard, for North America

1 – Australia

2 – United Kingdom

3 – United Kingdom

4 – China

5-7 – Not used

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 42: LD 15 - Modify Customer Data Block to introduce Alternative Pad Matrix 4 for China.

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

Feature operation

No specific operating procedures are required to use this feature.

Chapter 19: Alternative Loss Plan

Contents

This section contains information on the following topics:

Feature description

Customers can insert or remove, during administration, an alternative trunk-pad switching matrix using this feature. The loss-plan requirements of different countries can thus be satisfied. The alternative fixed trunk-pad matrix can be used in place of the standard pad switching matrix. See Figure 3: Alternative Loss Plan pad switching matrix on page 185.

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

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

Operating parameters

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

Feature interactions

B34 Codec Static Loss Plan Downloading

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

B34 Dynamic Loss Switching

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

R2MFC 1.5 Mbps Digital Trunk Interface

Alternative Loss Plan is not supported on 1.5 Mbps DTI.

Feature packaging

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

Feature implementation

Task summary list

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

1. Table 43: LD 14 - Configure the Trunks. on page 185
2. Table 44: LD 15 - Configure the Alternative Pad Matrix. on page 185

Table 43: LD 14 - Configure the Trunks.

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

Table 44: LD 15 - Configure the Alternative Pad Matrix.

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

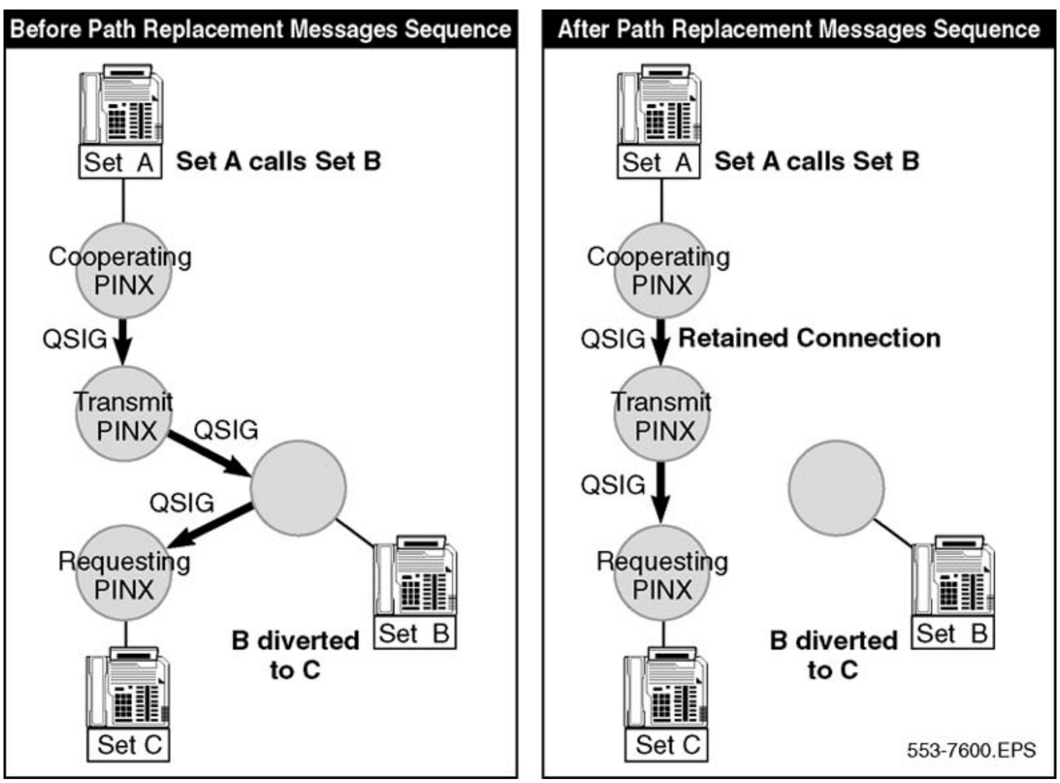


Figure 3: Alternative Loss Plan pad switching matrix

Feature operation

No specific operating procedures are required to use this feature.

Chapter 20: Alternative Routing for DID/DDD

Contents

This section contains information on the following topics:

Feature description

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

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

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

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

Scenario 1

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

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

Scenario 2

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

Scenario 3

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

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

Note:

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

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

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

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

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

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

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

Feature packaging

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

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- New Flexible Code Restriction (NFCR) package 49
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Pretranslation (PXLT) package 92
- Incoming Digit Conversion (IDC) package 113
- Integrated Digital Access (IDA) package 122
- Digital Private Network Signaling System 1 (DPNSS) package 123
- Digital Access Signaling System 2 (DASS2) package 124

Feature implementation

Table 45: LD 90 - Assign an ARRN and ARLI to an SPN.

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

Feature operation

No specific operating procedures are required to use this feature.

Chapter 21: Application Module Link

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

Chapter 22: Application Module

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

Chapter 23: Attendant Administration

Contents

This section contains information on the following topics:

Feature Description

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

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

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

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

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

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

For details on how these features operate, see Attendant Administration User Guide.

Operating Parameters

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

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

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

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

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

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

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

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

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

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

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

Feature Interactions

Attendant Administration does not support the following features:

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

Attendant Consoles

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

Attendant Position Busy

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

Attendant Supervisory Console

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

Automatic Wake Up

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

Call Forward No Answer/Flexible Call Forward No Answer

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

Call Hold, Deluxe

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

Console Presentation Group Level Services

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

Controlled Class of Service, Enhanced

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

Directory Number Delayed Ringing

The Attendant Administration feature is not supported.

End-to-End Signaling

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

Hot Line

Use of an attendant console to change the database for Enhanced Hot Line is not supported.

ISDN Calling Line Identification Enhancements

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

Multiple Appearance Directory Number Redirection Prime

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

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

CSC102 DN nnnn NEW MARP l s c u

where

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

Multi-Party Operations

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

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

Phantom Terminal Numbers (TNs)

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

Remote Call Forward

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

Speed Call, System

System Speed Call lists can be assigned using Attendant Administration.

Feature Packaging

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

Feature Implementation

Task summary list

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

• Table 46: LD 15 - Assign an Attendant Administration access code. on page 201

• Table 47: LD 12 - Add or change Attendant Administration key. on page 201

Table 46: LD 15 - Assign an Attendant Administration access code.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	PWD	Customer related passwords
CUST:		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E System.
- ATAC	xxxx	New or changed Attendant Administration access code (maximum four digits).
- - PWD2	xxxx	This password is programmed in LD 17 at the PWD2 prompt.

Table 47: LD 12 - Add or change Attendant Administration key.

Prompt	Response	Description
REQ	NEW	Add new data. Change existing data.
	CHG	
TYPE	2250	Attendant console type.
CUST	xx	Customer number, as defined in LD 15
TN		Terminal Number
	l s c u	Format for Large System, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx PRG	Add an Attendant Administration key.

Feature Operation

For details on feature operation, see Attendant Administration User Guide.

Chapter 24: Attendant Alternative Answering

Contents

This section contains information on the following topics:

Feature description

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

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

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

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

3. After the alternate telephone has been reached, the attendant console releases the call.
4. If call termination is unsuccessful at the AAA DN, an error message is generated that explains the problem:
    - If the error is because of an invalid AAA DN or tenant-to-tenant access denied condition, the call remains on the idle loop key for the attendant, and the AAA timer is not started again.
    - For all other errors, the call remains on the attendant loop key and the AAA timer is restarted. The sequence is repeated until the call is answered at the console, disconnected by the caller, or terminated at the AAA DN.

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

1. The AWU key buzzes, and the associated indicator fast flashes.
2. The attendant presses the AWU key to accept the recall.
3. The attendant presses the RLS key to release the call. An AWU recall must be acknowledged before any other calls can be presented to the attendant.
4. With AAA, the AWU call is presented to the attendant for the duration of the AAA timer. If an AWU recall is not acknowledged before the timer threshold, the recall is returned to the attendant queue to be presented later. The AWU recall will not be forwarded to the AAA DN.

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

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

1. Call Forward All Calls
2. Message Center
3. Call Forward No Answer
    - Flexible Call Forward No Answer
    - Second Level Call Forward No Answer
    - Call Forward by Call Type
4. Automatic Timed Recalls (slow answer)

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

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

1. Call Forward All Calls
2. Hunting
3. Call Waiting
4. Message Waiting (Direct Inward Dialing [DID] calls only) (if Message Waiting Forward Busy (MWFB) is enabled in LD 15)
5. Call Forward Busy (DID calls only)

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

Operating parameters

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

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

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

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

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

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

Feature interactions

Attendant Overflow Position

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

Attendant Recall

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

Call Forward All Calls

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

Call Forward Busy

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

Call Forward by Call Type

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

Call Forward No Answer

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

Call Pickup

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

Centralized Attendant Service

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

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

Do Not Disturb

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

Group Hunt

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

Hunting

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

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

Manual Line Service

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

Message Center

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

Multi-Tenant Service

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

Feature packaging

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

Feature implementation

Task summary list

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

1. Table 48: LD 15 - Configure the Attendant Alternate Answering feature. on page 208
2. Table 49: LD 12 - Define the AAA DN for each attendant console affected. on page 209

Table 48: LD 15 - Configure the Attendant Alternate Answering feature.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console option.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- ATIM	(0)-126	AAA timer in two-second increments. Odd numbers are rounded down. ATIM = 0 disables the feature

Table 49: LD 12 - Define the AAA DN for each attendant console affected.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
AADN	xxx...x	Attendant Alternative Answering DN.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 25: Attendant and Network-Wide Remote Call Forward

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

Chapter 26: Attendant Announcement

Contents

This section contains information on the following topics:

Feature description

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

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

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

Attendant Announcement types

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

Configure the following announcement types in LD 56:

- Announcement when terminating to the Attendant (ANAT)

  When a call is dialed directly, or intercepted, to the attendant, the caller receives an ANAT announcement.
- Announcement when Night Service is activated (ANNS)

  When a call is terminated to the night station or the night service queue, the caller receives an ANNS announcement.
- Announcement when Call Forward No Answer to the Attendant (ANFA)

  When a Call Forward No Answer (CFNA) call is redirected to the attendant, the caller receives an ANFA announcement.
- Announcement when Call Forward Busy to the Attendant (ANFB)

  When a Call Forward Busy (CFB) or Hunt call is redirected to the attendant, the caller receives an ANFB announcement.
- Announcement when Slow Answer Recall to the Attendant (ANSR)

  When a call is extended by the attendant and the call is not answered within Recall Timer (RTIM) time, the caller is redirected to the attendant and receives an ANSR announcement.
- Announcement on Attendant Extended Calls (ANXC)

  When an attendant transfers a trunk call to an extension, the caller receives an ANXC announcement until the requested party goes off hook.
- Announcement when Overflowed or Forwarded (ANOF)

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

Table 50: Announcement received at termination. on page 214 summarizes the types of announcements provided to the caller when a call is terminated to the attendant, attendant queue, night station or night queue.

Table 50: Announcement received at termination.

Type of call	Call destination	Announcement received
Direct calls or abandoned calls	Attendant or attendant queue	ANAT
Direct calls or abandoned calls	Night station or night queue	ANNS
Call Forward No Answer treatment (CFNA)	Attendant, attendant queue, or night station	ANFA
Call Forward Busy treatment (CFB)	Attendant, attendant queue, or night station	ANFB

Type of call	Call destination	Announcement received
Slow Answer Recall	Attendant, attendant queue, or night station	ANSR
Attendant Extended calls	Attendant, attendant queue, or night station	ANXC
Overflowed or Forwarded calls	Attendant, attendant queue, or night station	ANOF

Special options

During normal operations, when a call terminates to the night station, the ANNS announcement is given. This also applies to redirected calls. If the Night station announcement priority (NIPR) option in the announcement profile is set to "NO", calls redirected to the night station receive an appropriate greeting.

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

Announcement source types

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

- Tone interface
- Recorded Announcement (RAN) trunk interface (for example, Avaya Integrated Recorded Announcer)

Tone interface announcements

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

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

Tone interface announcements are configured in LD 56.

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

RAN trunk interface announcements

Attendant announcements can be provided by existing RAN trunks.

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

Answer Supervision for RAN trunks

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

Post-RAN post treatment

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

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

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

Alternative Attendant Announcement treatment

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

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

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

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

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

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

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

Attendant Alternative Answer

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

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

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

No Call Answer

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

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

Call Answer on Announcement

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

An answer is provided in the following cases:

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

Call Answer Forced

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

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

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

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

Operating parameters

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

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

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

Attendant Announcement is not provided on series calls.

Feature interactions

Attendant Alternative Answering

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

Attendant Barge-In

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

Attendant Clearing during Night Service

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

Attendant Forward No Answer

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

Attendant Interpositional Transfer

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

Attendant Overflow Position

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

Attendant Recall

Attendant Announcement does not support the Attendant Recall feature.

Automatic Call Distribution

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

Automatic Timed Reminders

An Automatic Timed Reminders recall receives the appropriate announcements.

Call Detail Recording Time to Answer

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

Call Forward All Calls

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

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

Call Forward No Answer Call Forward Busy Slow Answer Recall

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

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

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

Centralized Attendant Service

Centralized Attendant Service does not support Attendant Announcement.

DPNSS1

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

Direct Inward Dialing Call Forward No Answer Timer

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

EuroISDN Connected Number

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

MCDN-QSIG Gateway

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

Trunk Anti-Tromboning

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

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

Recorded Overflow Announcement

Attendant Announcement takes precedence over Recorded Overflow Announcement.

Slow Answer Recall

Slow Answer Recall calls receive ANSR announcement when specified.

Virtual Network Service

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

Feature packaging

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

This feature also requires the following existing packages:

- Recorded Announcement (RAN) package 7 (if RAN Announcements are used)
- Attendant Overflow Position (AOP) package 56 (if AOP is used)
- Flexible Tones and Cadences (FTC) package 125
- Attendant Forward No Answer (AFNA) package 134 (if AFNA is used)
- Network Attendant Service (NAS) package 159 (if used over MCDN network)
- Message Intercept (MINT) package 163
- Attendant Alternative Answering (AAA) package 174 (if AAA is used)
- Recorded Announcement Broadcast (RANBRD) package 327 (if the broadcast facility of the RAN trunk is used)

Feature implementation

Task summary list

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

1. Table 51: LD 56 — Configure the Attendant Announcement table on page 223
2. Table 52: LD 56 - Configure tone announcement for Large Systems. on page 224
3. Table 53: LD 16 - Enable the Attendant Announcement. on page 224

Use the following to configure announcements provided by RAN services:

1. Table 54: LD 16 — Configure RAN routes for Attendant Announcement on page 225
2. Table 55: LD 56 - Configure the Attendant Announcement table for RAN usage. on page 226
3. Table 56: LD 16 - Configure Route Data Block for Attendant Announcement. on page 227

Announcements provided by XCT/TDS tone service

Table 51: LD 56 — Configure the Attendant Announcement table

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

Prompt	Response	Description
		aaa = SRC1 - SRC8 source entry of the appropriate tone table.

Table 52: LD 56 - Configure tone announcement for Large Systems.

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

Table 53: LD 16 - Enable the Attendant Announcement.

Prompt	Response	Description
REQ	NEW CHG	Add a new data. Change existing data.
TYPE	RDB	Route Data Block.

Prompt	Response	Description
...		
TKTP		Trunk Type
	a..a	Attendant Announcement is available on DID, TIE and COT trunks only.
ATAN		Attendant Announcement.
	(NO) YES PSTN	No Attendant Announcement. Enable Attendant Announcement on this route. Enable Attendant Announcement on this route for PSTN calls only (for MCDN trunks only).
- ATBL	xx	Announcement profile table, where: xx = 0-31 This number should correspond with what you set at the AANN prompt to in LD 56.
- AAT		Alternative Attendant Announcement.
	(NO) YES	Disable Alternative Attendant Announcement. Enable Alternative Attendant Announcement.
--AATO	(0) - 3	Alternative Attendant Announcement Time of Day option.
--ADAY	(0) - 3	Alternative Attendant Announcement Day of Week option.
--AHOL	(0) - 3	Alternative Attendant Announcement Holiday option.
--AATB	xx	Announcement Profile Table for Alternative Announcement, where: xx = 0-31 This number should correspond with what you set at the AANN prompt to in LD 56.
- AAAO		Attendant Alternative Answer Option.
		This option is for Tone Announcements only.
	(NO) CAA CAF	No call answer is given Call answer will be given on announcement. Call answer will be given forced.

Announcements provided by RAN services

Table 54: LD 16 — Configure RAN routes for Attendant Announcement

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
TKTP		Trunk Type.
	RAN	RAN route.
RTYP	MCON	Continuous multichannel.
REP	1-15	Number of repetitions of this RAN.
STRT	DDL	Delay call connection until start of announcement.
BDCT		Broadcast Capability.
	(NO) YES	Deny RAN Broadcast Capability for this route. Allows RAN Broadcast Capability for this route.
WAIT	RGB	Provide ringback tone for calls queuing for RAN trunk.
ASUP		Answer Supervision.
	(NO)	Answer Supervision is controlled in the RDB of the incoming trunk route.
	YES	Return Answer Supervision.
RANH		RAN or Music route to use after post treatment.
	0-511	Range for Large System and Avaya Communication Server 1000E system.

Table 55: LD 56 - Configure the Attendant Announcement table for RAN usage.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AANN	Attendant Announcement data block.
CUST	xx	Customer number, as defined in LD 15
TBL	0-31	Announcement table number.
- NIPR		Night station Announcement Priority.
	(NO) YES	ANNS is not provided on each call to the night station. ANNS is provided on each call to the night station.
- ANQU		Attendant Queue.
	(NO)	Announcement is not provided on calls in the attendant queue or night service queue only.

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

Table 56: LD 16 - Configure Route Data Block for Attendant Announcement.

Prompt	Response	Description
REQ	NEW CHG	Add a new data. Change existing data.
TYPE	RDB	Route Data Block.
...		
TKTP		Trunk Type.
	a..a	Attendant announcement is available on DID, TIE and COT trunks only.
ATAN		Attendant Announcement.
	(NO) YES PSTN	No Attendant Announcement. Enable Attendant Announcement on this route. Enable Attendant Announcement on this route on PSTN calls only (For MCDN trunks only).

Prompt	Response	Description
- ATBL	xx	Announcement profile table, where: xx = 0-31 This number should correspond with what you set the AANN prompt to in LD 56.
- AAT		Alternative Attendant Announcement.
	(NO) YES	Disable Alternative Attendant Announcement. Enable Alternative Attendant Announcement.
--AATO	(0) - 3	Alternative Attendant Announcement Time of Day option.
--ADAY	(0) - 3	Alternative Attendant Announcement Day of Week option.
--AHOL	(0) - 3	Alternative Attendant Announcement Holiday option.
--AATB	xx	Announcement Profile Table for Alternative Announcement, where: xx = 0-31 This number should correspond with what you set at the AANN prompt to in LD 56.
- AAAO	(NO)	No Call Answer is given (for Tone Announcement only).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 27: Attendant Barge-In

Contents

This section contains information on the following topics:

Feature description

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

Operating parameters

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

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

The system must be equipped with a conference loop.

Feature interactions

Automatic Redial

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

Call Forward/Hunt Override Via Flexible Feature Code

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

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

Call Page Network Wide

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

Charge Account and Calling Party Name

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

China - Attendant Monitor

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

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

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

Conference

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

End-to-End Signaling

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

Intercept Computer Dial from Directory - Pre-dial Operations

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

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

ISDN Semi Permanent Connections for Australia

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

Uninterrupted Line Connections

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

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 57: LD 12 - Add or change a Barge-In key on attendant consoles. on page 232
2. Table 58: LD 10 - Allow or deny a warning tone Class of Service for analog (500/2500 type) telephones. on page 232
3. Table 59: LD 11 - Allow or deny a warning tone Class of Service for proprietary telephones. on page 233
4. Table 60: LD 14 - Allow or deny warning tone Class of Service for trunks. on page 233

Table 57: LD 12 - Add or change a Barge-In key on attendant consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	1 BIN	Add a Barge-In key.

Table 58: LD 10 - Allow or deny a warning tone Class of Service for analog (500/2500 type) telephones.

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

Prompt	Response	Description
CLS	(WTA) WTD	(Allow) deny warning tone.

Table 59: LD 11 - Allow or deny a warning tone Class of Service for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	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, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(WTA) WTD	(Allow) deny warning tone.

Table 60: LD 14 - Allow or deny warning tone Class of Service for trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	COT DID FEX RAN TIE WATS	Trunk type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(WTA) WTD	(Allow) deny warning tone.

Feature operation

To establish a connection on a trunk, follow these steps:

1. Select an idle loop key.
2. Press Barge-In.
3. Dial the route access code and the trunk member number, followed by the octothorpe (#).

The possible results are the following:

- dial tone (trunk is idle and working)
- conversation (trunk is busy and working)
- modem carrier tone (long distance trunk is working)
- fast busy (trunk is either disabled or has Warning Tone Denied Class of Service)

If you hear fast busy, check the trunk again before reporting a problem.

Chapter 28: Attendant Break-In

Contents

This section contains information on the following topics:

Feature description

The Attendant Break-In (BKI) feature simplifies the process required if an attendant must break in to an established call. When an attendant receives an urgent call and dials the destination DN, that DN may be busy. The attendant may then have to break in to the call. This feature provides a new key on the attendant console: the Break-In key. This feature allows the attendant to extend a call to a busy extension through a simple key operation.

The break-in process involves the following steps:

1. Use the Break-In key upon receiving the busy tone. This action establishes a conference between the attendant and the members of the established call (but excludes the incoming call). Parties hear the intrusion tone; secrecy is preserved.
2. Announce that an important call is waiting and request that the other parties disconnect from the call.
3. Extend the incoming call to the previously busy DN as soon as the other parties disconnect.

By using the Break-In key before dialing the destination DN, the attendant can override features such as Call Forward and Hunting.

Operating parameters

The Attendant Break-In feature is supported on analog (500/2500 type) telephones and proprietary telephones.

A console can have only one Break-In key.

A break-in connection cannot be put on hold.

Only one attendant at a time can break in to a call.

Attendant Break-In does not operate in the following situations:

- A party to the established call has Override Denied or Warning Tone Denied Class of Service
- The established call involves any of the following:
    - An attendant
    - Multi-frequency Compelled (MFC) device type
    - Digitone Receiver (DTR) device type
    - Page trunk
    - Dictation trunk
    - Recorded Announcement trunk
    - Integrated Voice and Message System (IVMS)
- The destination DN is on an outgoing trunk call. If the station is involved with an outgoing trunk call, the call is established when End of Dialing (EOD) times out, the number is dialed, or the trunk is answered.

Feature interactions

Attendant Blocking of Directory Number

The Attendant Blocking of DN and the source side Predial Break-in features are mutually exclusive for the same call. If the SACP key lamp is lit when the Break-in key is pressed to start a Predial Break-in attempt, the Break-in key is ignored. On the contrary, if the Break-in key lamp is lit and no call attempt is made on the source side when the SACP key is pressed to start an Attendant Blocking of DN, the SACP key is ignored.

If a Break-in attempt is made for an Attendant Blocking of DN call, the Break-in attempt will be considered to be temporarily denied.

It will be possible to Break-in on the destination side with an Attendant Blocking of DN call on the source side of the attendant console. The same limitations to Break-in will apply as if the source side call is a normal call.

Attendant Break-In to Inquiry Calls

All other interactions are the same as for the Attendant Break-In feature.

Attendant Busy Verify

The attendant can use the Break-In key instead of Busy Verify to break in to an established call. Attendant Break-In simplifies this process.

Automatic Call Distribution

Once the destination DN has established the call with the Automatic Call Distribution (ACD) agent, the attendant can break in to the call. If the destination DN is in the ACD queue, Attendant Break-In is temporarily denied.

Automatic Redial

Attendant Break-In and Attendant Busy Verify are not permitted on a proprietary telephone that is used for an Automatic Redial (ARDL) call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Busy Verify on Calling Party Control Call

Local Attendant Break-In will be temporarily denied if the desired party is already in a toll operator Break-In conference or on a Special Service call, or awaiting the Special Operator signal. Local attendant/toll operator Break-In will be temporarily denied if the desired party is established on an incoming toll call.

Call Forward All Calls

By pressing the Break-In key before dialing the destination DN, the attendant can override call forwarding on the destination DN. The attendant may not apply Camp-On to a telephone with Call Forward active.

Call Forward/Hunt Override Via Flexible Feature Code

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

The use of post-dial Break-in after dialing the Call Forward/Hunt Override FFC is possible after encountering a busy telephone, if Break-in is enabled.

Call Forward, Break-In and Hunt Internal and External Network Wide

If the Internal/External definition in LD 15 is set to YES, a call is treated as internal or external on a network wide basis.

Call Hold, Permanent Call Park

The attendant cannot break in to a call on hold or a parked call.

Call Page Network Wide

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

Call Transfer

The attendant cannot break in to a call that is being transferred until the transferred call is connected.

Call Waiting Camp-On

If the destination DN has a camped-on incoming trunk call, the attendant cannot extend the urgent incoming call as a Camp-On call.

Camp-on, Forced

Telephones with a toll operator break-in call cannot be camped on to. Overflow tone is returned to telephones attempting Forced Camp-on.

China - Attendant Monitor

If an attendant is monitoring a DN, a second attendant defined at the same customer site will be blocked from Breaking In to any party involved in the monitored call.

If an attendant is in a Break-In situation with a DN, a second attendant defined at the same customer site will be blocked from monitoring any party involved in the monitored call.

China Number 1 Signaling - Called Party Control

Attendant Break-In is not allowed on an outgoing Called Party Control call.

Conference

If the attendant cannot break in to a conference call because the call is supporting the maximum number of callers, busy tone continues and the Break-In key lamp flashes.

Digit Display

During Attendant Break-In, the Attendant Console Digit Display shows the DN of the incoming call and the destination DN until the attendant extends the incoming call to the destination DN and releases the connection.

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Group Hunt

Attendant Break-in will not be supported when dialing a Pilot DN directly.

Hold

The attendant cannot break in to a call on hold.

Hunting

If the destination DN is in a Hunting chain with some idle DNs, the Break-In request goes to the first idle DN in the chain. To prevent this occurrence, the attendant can press the Break-In key prior to dialing the destination DN.

Intercept Computer Dial from Directory - Post-dial Operation

Attendant Break-in

An attendant can break-in to a call by:

- Dialing an extension DN from the Intercept Computer.
- Pressing the Break-in key on the attendant console.

Make Set Busy Do Not Disturb

For a telephone with Make Set Busy or Do No Disturb in effect, Break-In is temporarily denied to the attendant. The Break-In lamp uses slow flash to indicate this situation. Using the Break-In key prior to dialing the destination DN circumvents this situation. After the Break-In, the telephone returns to its prior status.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Make Set Busy that may be applied to the telephone.

Meridian 911 Call Abandon

Since an abandoned call does not have a speech path established, the Break-In deny treatment is given to the attendant so that Break-In cannot occur.

Multiple Appearance Directory Number Redirection Prime

The attendant may get a busy tone if all the telephones with the required DN are busy. Break-In permits the attendant to break in to the connection with the least restricted TN. Where more than one TN exists that meets this criterion, Break-In chooses the one at the bottom of the DN block.

Multi-Party Operations - Three-Party Service

Break-In is not allowed to the party receiving the patience tone or the misoperation ringback.

Multi-Party Operations Enhancements

Attendant Break-in is not allowed to a connection in which a party is receiving Patience Tone or recall of misoperation ringback.

On Hold on Loudspeaker

It will not be possible to Break-in into a call on loudspeaker as it is effectively on hold at the telephone.

Override

When one telephone has overridden an existing call to establish a conference call, Break-In is temporarily denied. The attendant is notified by the override tone.

Priority Override

Telephones with a toll operator break-in call cannot be overridden. Overflow tone is returned to telephones attempting Priority Override.

Override, Enhanced

Telephones with a toll operator break-in call cannot be camped on to or overridden. Overflow tone is returned to telephones attempting either Forced Camp-on or Priority Override.

Periodic Camp-on Tone

The Periodic Camp-On Tone has precedence over Break-In intrusion tone.

Semi-Automatic Camp-On

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

Source Included when Attendant Dials

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

Trunk Barring

Trunk Barring does not result in intercept treatment for Toll Operator Break-In.

Feature packaging

Attendant Break-In (BKI) is package 127.

Feature implementation

Table 61: LD 12 - Assign the Break-In key on the attendant console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx BKI	Break-In key.

Feature operation

The operator can press the Break-In key either before or after dialing the destination DN. Break-In operates slightly differently in these two situations, as described below.

Post-Dial Break-In

For post-dial break-in

1. The attendant answers an incoming external call.
2. The attendant dials the destination DN.
3. The attendant receives the busy tone (unless the destination DN allows Camp-On or Call Waiting).
4. The attendant presses the Break-In key.
5. If allowed, the attendant joins the call on the destination DN to announce the incoming call and request that other parties disconnect. (See Table 62: Attendant console break-in states on page 244 for an explanation of console break in states.)
6. After the other parties disconnect, the attendant extends the incoming call to the destination DN.

Pre-Dial Break-In

For pre-dial break-in

1. The attendant answers an incoming external call.
2. The attendant presses the Break-In key.
3. The attendant dials the destination DN.
4. If the destination DN is busy, the attendant hears the busy tone; processing is the same as for Post-Dial Break-In above.

   If the destination DN is not busy, the DEST lamp flashes and the Break-In lamp goes dark. The attendant hears the ringback tone. Pressing the Break-In key a second time causes normal call processing for an idle line.

   If the destination DN is invalid, the attendant hears the overflow tone and the Break-In lamp goes off. To return to the source call, the attendant presses the Release Destination key.

Table 62: Attendant console break-in states on page 244 describes the possible attendant console break in states. These states depend on several factors:

- whether the source call is an external call
- the type of call in effect at the destination DN
- the combination of features allowed at the destination DN
- whether the attendant pressed the Break-In key before or after dialing the destination DN

Table 62: Attendant console break-in states

Console State	Lamp State	Description
ALLOW	Destination = LIT Break-In = LIT Tone = INTRUSION	The attendant can break in to the call and extend the incoming call.
CONSULT ONLY	Destination = FLASH Break-In = LIT Tone = INTRUSION	The attendant can break in to the call but cannot extend the incoming call.
TEMPORARILY DENIED 1	Destination = FLASH Break-In = FLASH Tone = BUSY/ OVERRIDE	The attendant temporarily cannot break in to the call, and may attempt the break in later.
TEMPORARILY DENIED 2	Destination = FAST FLASH Break-In =- FLASH Tone = OVERFLOW	The attendant temporarily cannot break in to the call.

Console State	Lamp State	Description
DENIED	Destination = FLASH Break-In = DARK Tone = OVERFLOW	The attendant cannot break in to the established call or extend the incoming call.
BREAK-IN IGNORED	Destination = FLASH Break-In = DARK Tone = RING BACK	The attendant cannot break in. The attendant should make a second break in attempt.
INVALID DN	Destination = FLASH Break-In = DARK Tone = OVERFLOW	The attendant attempted to reach an invalid DN. The attendant should dial the correct destination DN.

Chapter 29: Attendant Break-In Busy Indication and Prevention

Contents

This section contains information on the following topics:

Feature description

This feature, operating either in a standalone or Integrated Services Digital Network (ISDN) environment, provides enhancements to the Attendant Break-in feature. For more information, see *ISDN Primary Rate Interface Features Fundamentals* (NN43001-569).

Break-in Busy Indication

If an attendant, during a break-in operation, dials a busy extension, the attendant console display provides one of the following customer-defined indications:

- three dashes, appended to the end of a digit display (if the busy station is involved in an external call)
- a mode digit, appended to the end of a digit display

In a non-ISDN environment, the mode digit indicates one of the states:

1 = Station is busy on an external call, or station is busy on an off-net call.

2 = Station is busy on an internal call, or station is busy on an on-net call.

3 = Station is busy on a non-established call; for instance, dialing, ringing, or announcement. Or, station is busy on a conference call.

4 = Station is in line lockout.

In an ISDN Primary Rate Interface (PRI) environment, the mode digit indicates one of the following states:

1 = Station is busy on an off-net call, or involved in a conference call.

2 = Station is busy with on-net call, and is not involved in a conference call.

3 = Station is busy on a non-established call; for instance, dialing, ringing, or announcement.

4 = Station is in line lockout.

Break-in Prevention

A Break-in to External Call Denied (BIXD) option is provided to the customer which, if selected, temporarily denies Break-in to a party involved in an external call. This applies to both pre-dial and post-dial Break-in operations.

Operating parameters

The same limitations apply as for the Attendant Break-In and Network Attendant Service (NAS) Break-In features.

Feature interactions

All of the same feature interactions apply as for the Break-in and Network Attendant Service Break-in features.

The appropriate busy indication is given to a Line Lockout Set which has been broken in on.

Feature packaging

Attendant Break-In Busy Indication and Prevention requires Attendant Break-in/Trunk Offer (BKI) package 127.

Feature implementation

Table 63: LD 15 - Define break-in Indication and Prevention options.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT_DATA	Attendant console options.
...		
- OPT	(BIXA) BIXD (BIND) BBIN EBIN	Break-in to external call (allowed) denied. Break-in Indication (denied), Basic Beak-in Indication. Extended Break-in Indication.

Feature operation

For operating procedures, see Attendant Break-In Busy Indication and Prevention on page 247.

Chapter 30: Attendant Break-In to Inquiry Calls

Contents

This section contains information on the following topics:

Feature description

The Attendant Break-In to Inquiry Calls feature allows an attendant to Break-In to an inquiry call. An inquiry call exists when two stations are established in a simple connection, and one station offers a call transfer to another station. The telephone making the call transfer becomes the controlling party, and the station receiving the call transfer becomes the active party. The other station is placed on hold and becomes the held party.

The attendant can Break-In to either the controlling or active party, in post-dial or pre-dial operation, by pressing the Break-In (BKI) key. After Break-In has occurred, a Break-In conference is established. All parties receive intrusion tone. While in the Break-In conference, the attendant has consultation status only. The attendant cannot extend a call from the source side.

The attendant cannot Break-In to the held call, to an inquiry call that is in the dialing state or ringing state, or to the active or controlling party if either of them has Warning Tone Denied Class of Service.

To release from the Break-In conference, the attendant presses either the RLS SRC key (to release from source) or RLS DEST key (to release from destination). The inquiry call is restored to its previous state.

Operating parameters

Once in the Break-In conference, the operation of the console Release key is ignored. The operation of the Transfer key (TRN) and Add-on Conference key (AO3/AO6) for proprietary telephones is ignored. For analog (500/2500 type) telephones, a switchhook flash, ground button, or recall operation is ignored.

This feature does not allow the attendant to Break-In to a held party, controlling party while dialing, or the active party during ringing.

The attendant will be unable to Break-In on an inquiry call if either the controlling or active parties has a Warning Tone Denied (WTD) Class of Service.

Feature interactions

Attendant Break-In

All other interactions are the same as for the Attendant Break-In feature.

Attendant Break-In with Secrecy

Attendant Break-In with Secrecy interacts with Attendant Break-In to Inquiry Calls (BIEC) when the desired party has gone on-hook leaving an undesired party off-hook and excluded. BIEC has enhanced the existing BKI feature by giving overflow tone to the undesired party if it is a 500 type telephone (irrespective of whether the undesired party was involved in an inquiry call or not). BKIS does not change this operation for non-BKIS calls.

BKIS has a choice of options to be given to the undesired party if the desired party goes on-hook while the undesired party is excluded. These are taken from the AOCS options in the Customer Data Block. These options are not given to the undesired party if the undesired party has a call on hold, this only applies to analog (500/2500 type) telephones. The BIEC treatment of giving overflow tone is done instead so that the undesired party can be reconnected to the held party.

Therefore, it is quite possible for analog (500/2500 type) telephones and trunks to get different treatment depending on the circumstances.

The following is a list of treatments for different circumstances:

- Existing BKI BIEC disconnects undesired parties when the desired party goes on-hook, except for analog (500/2500 type) telephones where overflow is given. Therefore proprietary telephones and trunks are disconnected.
- BKIS will give either overflow, transfer to attendant, or disconnect treatment to analog (500/2500 type) telephones or trunks. Proprietary telephones are disconnected.

Automatic Call Distribution Agent/Supervisory Consultation Calls

A consultation Call from an Automatic Call Distribution (ACD) agent to the supervisor, invoked on the Supervisor key on the agent telephone, is not considered an inquiry call and is not affected by the Break-In to Inquiry Calls feature.

Automatic Hold

A consultation call on a proprietary telephone, using a second DN along with Automatic Hold, is not treated as an inquiry call. The consultation call may be broken-in to, but the call held on the first DN is not involved in the Break-In.

Call Forward All Calls/Call Forward No Answer/Call Forward by Call Type/Do Not Disturb

The operation of these features are overridden on a analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

Call Forward All Calls/Call Forward No Answer/Make Set Busy/Do Not Disturb

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Call Forward All Calls/Call Forward No Answer/Make Set Busy/Do Not Disturb that may be applied to the telephone.

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Do Not Disturb

The operation of Do Not Disturb is overridden on an analog (500/2500 type) telephone that has inadvertently been placed on-hook during a Break-In conference to allow it to be re-rung by the attendant.

If the controlling party goes on hook in a Break-In conference, and is being re-rung by the attendant, the ringing takes precedence over Do Not Disturb that may be applied to the telephone.

Held Call Clearing

Held Call Clearing takes precedence over Break-In to Inquiry Calls.

Misoperation During Transfer/Inquiry

Break-In to Inquiry Calls takes precedence over Misoperation During Transfer/Inquiry on a proprietary telephone inadvertently placed on-hook during a Break-In conference, for those cases where the misoperation treatment differs.

Feature packaging

Attendant Break-In to Inquiry Calls requires the Attendant Break-In/Trunk Offer (BKI) package 127.

Feature implementation

Table 64: LD 12 - Assign Break-In (BKI) to a console key.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx BKI	Key number; Break-In.

Feature operation

For operating procedures, see Attendant Break-In to Inquiry Calls on page 251.

Chapter 31: Attendant Break-In to Lockout Set Denied

Contents

This section contains information on the following topics:

Feature description

The Break-In to Lockout Set Denied (BKLS) enhancement provides an option to prevent an attendant from breaking in on a analog (500/2500 type) telephone that is in a line-lockout state. This feature is applied on a customer basis and has precedence over other line-lockout or Break-In functions.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

BKLS takes precedence over any other feature dealing with Break-In to a line lockout state.

Feature packaging

Attendant Break-In to Lockout Set Denied requires Attendant Break-In/Trunk Offer (BKI) package 127.

Feature implementation

Table 65: LD 15 - Allow or deny the Break-In to Line Lockout Set feature:

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT_DATA	Attendant console Option
- OPT	(BLA) BLD	Break-In to Line Lockout Set (allowed) denied.

Feature operation

For operating procedures, see Attendant Break-In to Lockout Set Denied on page 257.

Chapter 32: Attendant Break-In with Secrecy

Contents

This section contains information on the following topics:

Feature description

The Attendant Break-In with Secrecy (BKIS) feature enhances the capabilities of the Attendant Break-In feature. When a Break-In conference (attendant, desired party, and undesired party) is established and intrusion tone is provided, the attendant can press the Break-In (BKI) key again to exclude the undesired party and talk to the desired party without the intrusion tone.

BKIS applies to both pre-dial and post-dial Break-In operations. In a post-dial situation, the attendant dials the desired party before pressing the BKI key. Whereas in a predial case, the attendant presses the BKI key prior to dialing the digits of the desired party.

BKIS operates in a stand-alone environment and within a Meridian Customer Defined Network (MCDN) Integrated Services Digital Network (ISDN) environment.

In an MCDN ISDN environment, BKIS is an enhancement of Network Attendant Service (NAS) Break-In (BKI). For more information about Network Attendant Service Break-in, see *ISDN Primary Rate Interface Features Fundamentals* (NN43001-569-B1).

Operating parameters

The same feature requirements apply as for the Break-In feature.

Within an ISDN environment

- All conditions for NAS Break-In must be met.
- In order for this feature to operate correctly over the network, all nodes connected to the attendant must have Break-In software equipped.

In all cases, when displays are equipped, the information displayed is consistent with current operation (that is, when connected to only one party, the display shows the number and name, if equipped and configured, of that party, and when connected to more than one party, the display is blank).

Feature interactions

Other than the interactions described below, the feature interactions are the same as for the Break-In and NAS Break-In features.

Break-In to Enquiry Calls

Break-In with Secrecy interacts with Break-In to Enquiry Calls (BIEC) when the desired party has gone on-hook leaving an undesired party off-hook and excluded. BIEC has enhanced the existing BKI feature by giving overflow tone to the undesired party if it is a 500 type telephone (irrespective of whether the undesired party was involved in an enquiry call or not). BKIS does not change this operation for non-BKIS calls.

BKIS has a choice of options to be given to the undesired party if the desired party goes on-hook while the undesired party is excluded. These are taken from the AOCS options in the Customer Data Block. These options are not given to the undesired party if the undesired party has a call on hold. This only applies to analog (500/2500 type) telephones. The BIEC treatment of giving overflow tone is done instead so that the undesired party can be reconnected to the held party.

Therefore, it is possible for analog (500/2500 type) telephones and trunks to get different treatment depending on the circumstances.

The following is a list of treatments for different circumstances:

- Existing BKI BIEC disconnects undesired parties when the desired party goes on-hook, except for analog (500/2500 type) telephones where overflow is given. Therefore proprietary telephones and trunks are disconnected.
- BKIS will give either overflow, transfer to attendant, or disconnect treatment to analog (500/2500 type) telephones or trunks. Proprietary telephones are disconnected.

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

Executive Intrusion and Break-In are mutually exclusive. Pressing the BKI key will activate Break-In or Executive Intrusion. In addition, intrusion is not allowed into a Break-In conference.

Multi-Party Operation

For Multi-Party Operation (MPO), the operation of features, for example, going on-hook and releasing from a call during the BKIS conference between the attendant and the desired party, takes precedence over MPO operations for those cases where the treatment differs from that defined by the customer.

All network nodes must have MPO software, with identical Multiple-party Operation (MPO) options. Otherwise, MPO options in the desired party's node have precedence.

Pertaining to MPO options, if the undesired party is not located on the same node as the desired party, the undesired party is considered as an external party on the desired party node.

Music

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Network Attendant Service (NAS)

The BKIS feature operates in a networking environment with regard to the NAS Break-In feature operations and limitations. For more information about the Network Attendant Service (NAS) feature, see *ISDN Primary Rate Interface Features Fundamentals* (NN43001-569-B1).

Secrecy Enhancement

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

Feature packaging

Attendant Break-In with Secrecy requires the following packages:

- Attendant Break-In (BKI) package 127.
- In an MCDN ISDN environment, ISDN Basic (ISDN) package 145, ISDN Supplementary Features (ISDNS) package 161, and Network Attendant Service (NAS) package 159 are required.
- Multi-Party Operations (MPO) package 141 is optional. If used in an MCDN ISDN environment, all nodes must be equipped with the MPO package.

Feature implementation

Task summary list

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

1. Table 66: LD 12 - Assign Break-In to a key on the attendant console. on page 262
2. Table 67: LD 15 - Modify Multi-Party Operations data if MPO package 141 is equipped. on page 263

Table 66: LD 12 - Assign Break-In to a key on the attendant console.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	2250	Attendant console type.
AADN	...	
KEY	0-19 BKI	Key number assigned to Break-In.

Table 67: LD 15 - Modify Multi-Party Operations data if MPO package 141 is equipped.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	MPO	Multi-Party Options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
MPOP	(NO) YES	Multi-Party Operations.
- FMOP	(NO) YES	Flexible Misoperation Options.
- - AOCS	xxxyyy	All Other Cases, where: xxx is for internal calls and yyy or ATN is for external calls.
	AAR AAR	The transferring station is re-rung. If the transferring station fails to answer, the transferred station is routed to the attendant.
	ATN(ATN)	Attendant
	DAR	The transferring station is re-rung. If the transferring station fails to answer, the transferred station is disconnected.
	(DIS)DIS	Disconnect
	OVF	Overflow
	STD	Standard

Feature operation

Break-In to two-party connection

The following sections describe a post-dial Break-In. For pre-dial Break-In, Break-In is done on the Source of the attendant and there is no party A calling the attendant, but the BKIS operation is identical.

The scenario is the following:

Party A calls the attendant. The attendant calls party B who is talking to party C. The attendant presses the BKI key to intrude into the conversation. At this point, the attendant and both parties B and C are in conversation with intrusion tone provided, while party A is on HOLD (with music if EMUS, package 119, is equipped).

Break-In, Allowed

This situation will arise when party A is an external call and Camp-on or Call Waiting is possible at the wanted station B. At this point, the BKI, Exclude Source (EXCL SRC) and Exclude Destination (EXCL DEST) indicators are active (lamps are lit or Liquid Crystal Display [LCD] is on), and the following actions can occur:

Request the unwanted party to terminate

The attendant may request the unwanted party to terminate. A positive response will terminate the conference that included the attendant and intrusion tone. This is a current BKI operation.

Request the wanted party to terminate

The attendant may request the wanted party to terminate the call. The party disconnects, terminating the BKI conference. This is a current BKI operation.

Attendant presses Release Destination key

The attendant may press the RLS DEST key to release the call. This action terminates the conference and the original call is reestablished as it was prior to Break-In. The Source party A is connected to the Attendant. This is a current BKI operation.

Attendant presses Exclude Destination key

The attendant may press the EXCL DEST key to return to the incoming call. The intruded parties keep receiving the intrusion tone. This is a current BKI operation.

Attendant presses Release key

The attendant may press the Release (RLS) key to apply Camp-on. This is a current BKI operation.

Attendant presses Break-In key again

The BKIS feature allows the attendant to press the BKI key again in order to exclude the undesired party C (who continues to hear intrusion tone) and to talk directly to the desired party B without intrusion tone. The BKI indicator, which was active, flashes at 60 impulses per minute (ipm).

Important:

When the attendant presses the BKI key a second time with the Break-In conference excluded, is not activated (that is, if the Break-In conference is on the destination but the attendant is talking on the source, secrecy cannot be activated).

From this point, the following attendant operations can occur:

Attendant actions

Break-In

The attendant presses the flashing BKI key. In this case, party C, which was excluded, is brought back into conversation with the attendant, party B, and intrusion tone. The BKI indicator reverts to an active state. The situation reverts to a normal BKI conference with intrusion tone.

In other words, the lit BKI key can be used to exclude the unwanted party from the BKI conference and the flashing BKI key can be used to reestablish the BKI conference (with intrusion tone).

Exclude Destination

The attendant presses the EXCL DEST key to return to the incoming call. The attendant is connected to the source party. The unwanted party B and the wanted party C are reconnected with intrusion tone. The EXCL SRC indicator is now off and the EXCL DEST lamp and the BKI indicators are active. The operation of the EXCL DEST key has the same effect as for a normal BKI conference situation, as described previously.

Release

The attendant presses the RLS key to apply Camp-on. If Camp-on or Call Waiting is available, parties B and C are reconnected and party A is released and either Camp-on or Call Waiting is applied to the wanted party A. The BKI indicator is off. If Camp-on or Call Waiting is not available, the operation of the RLS key causes secrecy to be turned off and the situation to go back to the Break-In conference with intrusion tone. The loop can only be released by pressing the RLS DEST key, leaving the source connected to the attendant. The operation of the RLS key has the same effect as for a normal BKI conference situation, as described previously.

Release Destination

The attendant presses the RLS DEST key. The BKI, EXCL SRC, and EXCL DEST indicators are off and party A is connected to the attendant. Party B (desired) and party C (excluded party) are reconnected.

Undesired party action

Party C (undesired party) goes on-hook and is disconnected. Then the BKI indicator goes off and the attendant treats the call as a normal two-party connection. The attendant is talking directly to party B (desired party) and can press the RLS key to extend the call.

Desired party action

At this point, if party B (controlling party) goes on-hook, the treatment depends upon the Customer Data Block (LD 15) Multi-party Operations (MPO) Flexible Misoperation Options (FMOP) All Other Cases (AOCS) settings if the undesired party is a trunk or 500-type telephone and MPO package 141 is equipped. If the MPO package is not equipped, internal calls will be disconnected, while external calls will be rerouted to the attendant.

The following shows what happens to 500-type telephones or trunks depending on the AOCS options:

AOCS set to AAR for party C

If AOCS is set to AAR for party C, then party C is routed to the attendant and party B is re-rung by the attendant. BKI indicator goes off and a simple call is set up between attendant and party B when B answers.

AOCS set to ATN for party C

If AOCS is set to ATN for party C, then party C is routed to the attendant while B is re-rung by the attendant. The BKI indicator goes off and the attendant hears ring back and the DEST indicator winks at 30 ipm. The attendant can extend the call as normal.

AOCS set to DAR for party C

If AOCS is set to DAR for party C, then party C is disconnected and party B is re-rung by the attendant. The BKI indicator goes off and when B answers a simple call exists between the attendant and party B.

AOCS set to DIS for party C

If AOCS is set to DIS for party C, then C is disconnected and party B is re-rung by the attendant. The BKI indicator goes off and the attendant hears ringback and the DEST indicator winks at 30 ipm. The attendant can then extend the call as normal.

AOCS set to OVF for party C

If AOCS is set to OVF then overflow tone is given to party C and party B is re-rung by the attendant. The BKI indicator goes off, the attendant hears ringback, and the DEST indicator winks at 30 ipm. The attendant can then extend the call as normal.

AOCS set to STD for party C

If AOCS is set to STD for party C, the treatment is the same as default for the AOCS option. If party C is internal, then DIS option applies to party C, and if party C is external, then ATN option applies to party C.

Break-In (Consultation Only)

This console state indicates that the attendant has been allowed to Break-In to the desired party's call; however, the attendant will not be able to extend the originating call. This situation will occur under any of the following conditions:

- An internal call is on the source port of the attendant console.
- The attendant originated the call. In this case, the source indicator will be used instead of the destination indicator to provide status information (predial situation).
- An external call is on the source and neither Camp-on nor Call Waiting is possible at the wanted station (that is, Camp-on or Call Waiting not possible or the station already has a call camped on).
- The desired station is busy with Call Forward active and the attendant initiated a predial Break-In.

The BKI and the EXCL SRC indicators are active, the DEST indicator is flashing. At this point, the attendant is not allowed to press the RLS key to extend the originating call, party A. The operation of the RLS key is ignored. This is a current BKI operation.

The attendant may press the BKI key to exclude party C and talk directly to party B, as described under the Attendant actions section. The BKI and DEST indicators are flashing. While in this state, the attendant is not allowed to press the RLS key to extend the originating call, party A. The operation of the RLS key causes the secrecy to be turned off and the situation to revert to a Break-In conference. The other operations described in the Attendant actions section are available.

Break-In to a conference

Party A (either internal or external) calls the attendant, the attendant calls party B who is involved in a conference call with parties C and D. The attendant presses the BKI key to intrude into the conversation. At this point, the attendant, party B and all the original conferees are in conversation with intrusion tone provided, while party A is on HOLD. The BKI and EXCL SRC indicators are active. The DEST indicator is flashing and the BKI status is 'Consultation Only'.

At this point, the attendant may press the BKI key to talk directly to party B without intrusion tone. The Break-In indicator flashes at 60 ipm. The original conference is excluded from party B (the other parties in the conference remain connected without intrusion tone). Party A is still excluded on the attendant loop and the attendant is talking directly to party B without intrusion tone.

While in this state, the following situations can occur:

Attendant actions

Break-In

The attendant may press the flashing BKI key. The original conference is reestablished with intrusion tone. The BKI indicator reverts to active.

Exclude Destination

The attendant may press the EXCL DEST key to return to the incoming call. The original conference is reestablished and party A is connected to the attendant.

Release

The attendant is not allowed to extend the original call to the wanted party B by pressing the RLS key. The operation of the RLS key causes the secrecy to be turned off and the situation reverts to a Break-In conference.

Release Destination

The attendant may press the RLS DEST key. The BKI, EXCL SRC and EXCL DEST indicators are off and party A is reconnected to the attendant. The original conference (B, C, and D) is reestablished.

Undesired party action

All but one of the conferees (C or D) go on-hook. The last undesired party will start getting the intrusion tone once again. The situation reverts to the previously described operation (See Undesired party action on page 265).

Desired party action

At this point, if party B goes on-hook, party B is re-rung by the attendant and the conferees are left in conference without party B and without intrusion tone. The BKI indicator goes off, the attendant hears ringback tone, and the DEST indicator winks at 30 ipm. The attendant can extend the call as normal.

Table 68: Summary of possible Break-In situations and indications on page 268 is a summary of possible Break-In situations and indications.

Table 68: Summary of possible Break-In situations and indications

State	Operation	SRC or DEST Indicator	Break-In Indicator	Tone
1. Allowed	a) post-dial	ACTIVE	ACTIVE	intrusion
	predial	ACTIVE	ACTIVE	busy
	b) post-dial	ACTIVE	OFF	none
	predial	ACTIVE	ACTIVE->OFF	override
2. Consultation Only	a) post-dial	FLASH	ACTIVE	intrusion
	b) predial	FLASH	ACTIVE	busy
3. Temporarily Denied 1		FLASH	FLASH	busy override if override is involved
4. Temporarily Denied 2	a) post-dial only	FLASH	WINK	overflow
	b) predial	FLASH	WINK	busy or ring back
	(then post-dial)	FLASH	WINK	intrusion
5. Denied		FLASH	OFF	overflow
6. Break-In	a) post-dial	WINK	OFF	ringback
Ignored station is rung	b) Predial	WINK	OFF	ringback

State	Operation	SRC or DEST Indicator	Break-In Indicator	Tone
7. Invalid	post-dial or predial	OFF	OFF	overflow
8. Break-In with Secrecy	after post-dial or predial, active BKI key is pressed	ACTIVE or FLASH	FLASH	no tone

Table 69: Summary of possible Break-In situations and actions on page 269 is a summary of possible Break-In situations and actions.

Table 69: Summary of possible Break-In situations and actions

Condition of called DN	Action
1. Established call, Call Waiting or Camp-on allowed, Multiple Appearance DN. Lockout (if not denied).	Break-In allowed, connection established. Connection is made.
2. Attendant dialing on SRC, internal call on SRC, CWT or Camp-on not available, desired party in conference, Call Forward active on telephone.	Connection is made for the attendant only.
3. Tones, ringing, dialing, blocking, Override, Camp-on, Hold, talking to another attendant, Call Transfer, WTD on undesired party.	Release DEST, wait and repeat.
4. Make Set Busy, Do not disturb.	Predialing operation possible.
5. Warning tone denied on desired party, maintenance busy.	Break-In impossible.
6. Station is idle.	Station is rung, station not affected.
7. Invalid numbers.	Break-In impossible.
8. The previous status was "Allowed" or "Consultation Only". SRC or DEST indicator was active ("Allowed") or flashing ("Consultation Only").	Undesired party is excluded and the attendant is talking to the wanted party.

Chapter 33: Attendant Busy Verify

Contents

This section contains information on the following topics:

Feature description

Attendant Busy Verify allows the attendant to establish a connection with any apparently busy DN to verify that the DN is actually busy and in working order. This feature can also be used to connect with a busy station if an emergency situation requires call interruption by the attendant.

When Busy Verify is active, a 256 millisecond burst of interrupted tone is sent every six seconds to indicate the presence of the attendant. The attendant can Busy Verify only those stations with Warning Tone Allowed Class of Service.

When a station is involved in a conference, the attendant can verify whether the station is busy even if it has Warning Tone Denied Class of Service.

An attendant can also use either the Release Source or Release Destination key on the console to release one of the parties involved in a Busy Verify conference.

Operating parameters

The system must be equipped with a conference loop.

If equipped, the Busy Verify key must be assigned to key 0 of the console flexible feature strip.

Feature interactions

Attendant Break-In

The attendant can use the Break-In key instead of Busy Verify to break in to an established call. Attendant Break-In simplifies this process.

Automatic Redial

Attendant Break-In and Attendant Busy Verify are not permitted on proprietary telephones that are used for an Automatic Redial (ARDL) call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Call Forward All Calls

If the DN is call forwarded to the attendant console, the attendant will receive a click followed by silence.

Call Forward Busy Hunting

Call Forward Busy and Hunting do not affect Busy Verify.

Call Forward/Hunt Override Via Flexible Feature Code

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

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

Call Forward, Internal Calls

When the attendant is using this feature to call a telephone that is Internal CFW active, the call will not receive Internal CFW treatment.

Charge Account and Calling Party Number

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

China - Attendant Monitor

When China (CHINA) package 285 is equipped, the normal operation of Busy Verify changes. The repeatable tone is now configurable with the (TOA)/TOD option.

If an attendant is monitoring a DN, a second attendant defined for the same customer will be blocked from Busy Verifying any party involved in the monitored call.

If an attendant is Busy Verifying a DN, a second attendant defined for the same customer will be blocked from monitoring any party involved in the monitored call.

Conference

Conference Control cannot be activated if an attendant has used Busy Verify during a conference that involves a trunk.

Direct Inward System Access

Attendant Busy Verify applies only to DNs within the system. If an attendant tries to use the feature to enter a Direct Inward System Access DN, overflow tone is returned.

Group Hunt

An attendant is not allowed to busy-verify when dialing a Pilot DN directly.

Intercept Computer Dial from Directory - Pre-dial Operations

It is possible for an attendant to override call forward on a telephone in the following manner:

- Press an idle loop key, and press the Break-in key on the attendant console.
- Dial an extension DN from the Intercept Computer.

Music, Enhanced

When the attendant attempts to Busy Verify a telephone receiving Music, the Music is removed. When the attendant releases, Music is returned.

On Hold on Loudspeaker

It will not be possible to Busy Verify into a call on loudspeaker as it is effectively on hold at the telephone.

Periodic Camp-on Tone

The Periodic Camp-On Tone has precedence over Busy Verify intrusion tone.

Uninterrupted Line Connections

Attendant Busy Verify cannot be applied to stations with a Warning Tone Denied Class of Service.

Feature packaging

Attendant Busy Verify is included in base system software.

Feature implementation

Task summary list

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

1. Table 70: LD 12 - Add/change a Busy Verify key on attendant consoles. on page 275
2. Table 71: LD 10 - Allow/deny Warning Tone Class of Service for analog (500/2500 type) telephones. on page 275
3. Table 72: LD 11 - Allow/deny Warning Tone Class of Service for proprietary telephones. on page 276
4. Table 73: LD 14 - Allow/deny Warning Tone Class of Service for trunks. on page 276

Table 70: LD 12 - Add/change a Busy Verify key on attendant consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	0 BVR	Add a Busy Verify key.

Table 71: LD 10 - Allow/deny Warning Tone Class of Service for analog (500/2500 type) telephones.

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

Prompt	Response	Description
CLS	(WTA) WTD	Allow or deny warning tone.

Table 72: LD 11 - Allow/deny Warning Tone Class of Service for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	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, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(WTA) WTD	Allow or deny warning tone.

Table 73: LD 14 - Allow/deny Warning Tone Class of Service for trunks.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	COT DID FEX RAN TIE WAT	Trunk type.
TN		Terminal Number
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(WTA) WTD	Allow or deny warning tone.

Feature operation

To verify a busy DN, follow these steps:

1. Select an idle loop key.
2. Press **Busy Verify**.
3. Dial the DN of the station.

   If the DN is idle, press **Signal Source** to ring the station.

   Possible results are the following:

   - silence (DN is idle and working)

    - conversation (DN is busy and working)
    - fast busy (station is disabled or has Warning Tone Denied Class of Service).

4. Press the **Rls** key to disconnect from the call.

An enhancement to the Busy Verify feature offers the following functionality. Party A is on a call with Party B. The attendant

1. Selects an idle loop key.
2. Presses **Busy Verify**.
3. Dials Party A and creates a Busy Verify conference between Party A, Party B, and the attendant.

The use of the **Rls DEST** and **Rls SOURCE** keys are allowed at this point as follows:

- The attendant can press the **Rls DEST** key to release Party A from the Busy Verify conference or
- The attendant can press the **Rls SOURCE** key to release Party B from the Busy Verify conference.

Chapter 34: Attendant Call Selection

Contents

This section contains information on the following topics:

Feature description

All calls to the attendant, with the exception of slow-answer recalls, are automatically queued in order of arrival. The attendant can answer a call in two ways:

- Calls can be answered in the order received, regardless of call type, using the Loop key (LPK).
- A particular call type can be answered before other calls in the queue by manually selecting the appropriate Incoming Call Indicator (ICI) key.

The first call presented to an idle console is indicated by the appropriate ICI lamp. All subsequent calls are indicated by the Calls Waiting lamp only until the first call is released. All appropriate ICI lamps will then light, and an attendant may select a specific incoming call type by pressing the appropriate ICI key.

If a customer has multiple consoles, the first call in queue is presented to the first idle console.

Operating parameters

The maximum number of ICI lamps per attendant console is 20. All consoles associated with a customer have the same ICI assignments.

Feature interactions

Attendant Incoming Call Indicators

The ICI feature is used with the Attendant Call Selection feature to recognize, answer, and process incoming calls.

Feature packaging

This feature is included in base system software.

Feature implementation

No change to existing configuration is required for the Attendant Call Selection feature.

To implement ICI, see Attendant Incoming Call Indicators on page 280.

Feature operation

The attendant can answer a call by:

- pressing the Loop key to answer calls in the order received, or
- pressing the appropriate ICI key to answer a call by call type.

Chapter 35: Attendant Calls Waiting Indication

Contents

This section contains information on the following topics:

Feature description

Call Waiting on the console gives the attendant an indication of the number of calls in the console queue and the length of time they have been waiting to be answered. Each console is equipped with a Call Waiting indicator. The indicator is dark when no calls are waiting in the queue. The indicator is steadily lit when one or more calls are waiting. The indicator flashes when the number of waiting calls exceeds the customer defined threshold, or when a call has been waiting longer than the specified number of seconds.

The two thresholds that control the lamp states are defined in the Customer Data Block. The time delay threshold can be specified from 0 to 511 seconds in multiples of two seconds. The number of calls threshold can be specified from 0 to 255. If zero is specified, this aspect of the Call Waiting feature is not operational.

An option is also provided to supply a two-second buzz to notify the attendant when the first call enters the queue or when the Call Waiting lamp changes from steadily lit to flashing, or both.

If the threshold has been exceeded and the Call Waiting indicator is flashing, it changes to steadily lit when the threshold is no longer exceeded by either number of calls or time delay.

Operating parameters

If neither the time delay or number of calls thresholds are defined, the Call Waiting lamp state will not change from steadily lit to flashing.

Feature interactions

Call Park on Unsupervised Trunks

If all the attendants are busy and a Call Park Recall occurs, the recall is placed in the calls waiting queue. If the recalled station is busy when the recall occurs, the Disconnect Timer (DCTI) temporarily suspends timing until the recall is presented. After the recall is presented, the Disconnect Timer continues timing for the remainder of the period.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 74: LD 15 - Define Call Waiting thresholds and indications for a customer. on page 283
2. Table 75: LD 12 - Add/change a Display Calls Waiting key on an attendant console. on page 283

Table 74: LD 15 - Define Call Waiting thresholds and indications for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- CWUP	(NO) YES	Call Waiting Queue Update. (Do not) automatically notify Avaya 2250 Attendant Console when the number of calls waiting in queue changes.
- CWCL	(0)-255 (0)-255	Call Waiting Call Limit. Lower and upper bound of the threshold for the number of calls waiting (the default is 0).
- CWTM	(0)-511 (0)-511	Call Waiting Time. Lower and upper bound of the threshold for the time calls are waiting (the default is 0).
- CWBZ		Call Waiting Buzz
	(NO) YES	(Disable) enable a buzz to the attendant when either the CWCL or CWTM thresholds are exceeded.
	(NO) YES	(Disable) enable a buzz to the attendant when the first call enters the queue.

Table 75: LD 12 - Add/change a Display Calls Waiting key on an attendant console.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx DCW	Add a Display Calls Waiting key. xx = 00-19 for Avaya 2250 Attendant Console.

Feature operation

If CWUP (notify change in Calls Waiting status) is set to YES in LD 15, the number of calls waiting are displayed on the Avaya 2250 Attendant Console. If CWUP is set to NO, the attendant must press the Display Calls Waiting (DCW) key to display the number of waiting calls.

Chapter 36: Attendant Clearing during Night Service

Contents

This section contains information on the following topics:

Feature description

When an attendant console is placed in Night Service, the Attendant Clearing during Night Service feature causes all active calls or calls being held on Loop keys to be cleared and given a customer-defined treatment. One of the following treatments can be selected:

- internal calls are disconnected, and external calls are routed to the Night Directory Number (DN)
- all calls are routed to the Night DN
- no clearing

An external call is defined as a call involving at least one external party. The definition of a external party is the same as used for the Multi-Party Operations (MPO) feature. Any CO, DID, or TIE trunk (incoming or outgoing) connected to the system is considered an external party, regardless of the way the connection is established.

Operating parameters

Attendant Clearing during Night Service is offered as part of the Multi-Party Operations feature.

Feature interactions

AC15 Recall: Timed Reminder Recall

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

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

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

Night Service Enhancements

The Night Service Enhancements features take precedence over Attendant Clearing during Night Service.

Scheduled Access Restriction

Attendant Clearing during Night Service should be equipped with Scheduled Access Restriction (SAR). When Night Service is in effect, the only operations that can be performed from attendant consoles, which are members of a SAR group, are:

- release any existing calls, or
- dial one of the following SAR Flexible Feature Codes:
    - Scheduled Access Disable (SADS)
    - Scheduled Access Enable (SAEN)

- Scheduled Access Lock (SALK), or
- Scheduled Access Unlock (SAUN).

Feature packaging

The Attendant Clearing during Night Service feature is packaged as part of the Multi-Party Operations (MPO) package 141.

Feature implementation

Task summary list

The following task is required:

Table 76: LD 15 - Configure the Attendant Cleaning during Night Service feature at the ACNS prompt. on page 287

Table 76: LD 15 - Configure the Attendant Cleaning during Night Service feature at the ACNS prompt.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Modify existing data.
TYPE:	MPO	Multi-Party Options
...		
MPOP	YES	Multi-Party Operations options.
...		
- FMOP	YES	Flexible Misoperation Parameters.
- - ACNS		Attendant Clearing during Night Service.
	(NO) ALL EXT	Attendant will not be cleared (the default). All calls will be routed to the Night DN. External calls will be routed to the Night DN, while internal calls will be disconnected.
	<CR>	Previously defined value not changed, or set to default if response to REQ was NEW.

Important:

This overlay is modified to output the Attendant Clearing during Night Service (ACNS) prompt as part of the MPO group of prompts. The ACNS prompt only appears if the MPO package is equipped and the response to both MPOP and FMOP is YES. The ACNS prompt accepts a response of either NO, ALL, EXT or a carriage return (<CR>).

Feature operation

A customer is put into Night Service manually, by pressing the NITE key on the attendant console or having all attendant consoles activate Position Busy, or automatically, by the Scheduled Access Restrictions (SAR) or Attendant Forward No Answer (AFNA) features. When Night Service is activated, all calls or selected calls associated with the attendant will be given treatment according to the feature option defined in the Customer Data Block (LD 15) as part of the Multi-Party Operations (MPO) options.

The sections following describe the treatments given to different call types.

Established Calls

Single Party Call

Both the incoming or outgoing single party call, (not associated with another call on the attendant) established on the attendant Source (SRC) or Destination (DEST) sides will be routed to the Night DN.

Two Party Call - Ready to Extend

When a call is being extended, a call excluded on the SRC side and an outgoing call established on the DEST side, the call will be extended provided it is allowed as if the Release (RLS) key is pressed. If it is not allowed due to access restriction, the outgoing call on the DEST side will be disconnected and the call on the SRC side will be routed to the Night DN.

Conference Call on Source

If a conference call is established on the attendant SRC, the attendant will be excluded from the conference and disconnected as if the Release Source (RLS SRC) key were pressed.

Conference Call on Destination

If a conference call is established on the attendant DEST (Break-In conference) the attendant will be excluded from the conference and disconnected as if the Release Destination (RLS DEST) key were pressed.

Non-established Calls

Any call in the dialing state on either the SRC or DEST side will be disconnected.

Any call in the ringing state or receiving any tone on either the SRC or DEST side will be dropped or disconnected as if the RLS SRC or DEST key was pressed.

If the call in the ringing, dialing or receiving tone state is on the DEST side, and there is an established call in the EXCLUDE state on the SRC side, the SRC party will be rerouted to the Night DN.

Calls held on the console Loop keys

Any established calls being held on a Loop key will be released and calls extended where possible as described in the Established Calls section, or routed to the Night DN.

When a held call is routed to the Night DN, the held party, which is listening to silence or Music on Hold if available, will receive Ringback Tone. If the Night DN is not idle, the call will be placed in the Call Waiting queue.

Chapter 37: Attendant Consoles

Contents

This section contains information on the following topics:

Feature description

Attendant consoles assist in placing and extending calls into and out of the system. The operator of an attendant console is known as the attendant. The consoles provide the attendant with many unique features that increase the speed and ease of call processing.

This feature module provides an overview of attendant consoles and a description of the basic software capabilities and associated service changes. Additional information regarding attendant-related software features can be found in other feature modules in this document.

The Avaya 2250 Attendant Console is a Digital console with a 4-line, 40-character wide alphanumeric Liquid Crystal Display. The console's LCD displays the information presented in Table 77: LCD alphanumeric display information on page 291.

Table 77: LCD alphanumeric display information

Line	Display information
1	Displays the time and date.
2	Displays call source information.
3	Displays call destination information.
4	Displays console status information.

Directly below the display screen is a horizontal row of keys that provide the Position Busy, Night Service, Signal Source, and Signal Destination functions.

The Avaya 2250 Attendant Console has a digit display at the top of the console and a dial pad below the display. Five vertical keystrokes on the console provide access to the functions described in this section.

Vertical keystrip 1

This keystrip at the far left on the console is utilized for Trunk Group Busy (TGB) keys. The attendant can deny stations access to a trunk route by pressing the associated Trunk Group Busy key. Additionally, the lamps associated with Trunk Group Busy keys provide the visual indication of the status of the trunks within the route (See Table 78: Visual Indication of the status of the trunks within the route on page 292).

Table 78: Visual Indication of the status of the trunks within the route

Visual Indication	Status of the trunks within the route
Dark	Some of the trunks in the route are idle.
Flashing	All of the trunks in the route are busy.
Steadily lit	The attendant has taken control of the route.

The basic attendant console has 10 Trunk Group Busy keys. If an add-on module is installed, there are 16 Trunk Group Busy keys.

Vertical keystrip 2

This keystrip is used for Incoming Call Indicator keys. The Incoming Call Indicators (ICIs) identify the type of calls in the queue and the status of each particular call type. Three lamp states are associated with each Incoming Call Indicator key (See Table 79: Key lamp states associated with each Incoming Call Indicator key on page 292).

Table 79: Key lamp states associated with each Incoming Call Indicator key

Lamp state	Status of call type
Dark	No calls of this type are waiting.
Flashing	One call of this type is waiting in queue.
Steadily lit	Two or more calls of this type are queued, or one call has been waiting longer than 20 seconds.

To select a specific type of incoming call, the Incoming Call Indicator key associated with a steadily lit or flashing LED is pressed. The call is removed from the queue and presented to an idle loop key on the attendant console.

The basic attendant console has 10 Incoming Call Indicator keys. If an add-on module is equipped, the console may have 20 Incoming Call Indicator keys. An Incoming Call Indicator key may be assigned to one or more of the call types listed in Table 80: Incoming Call Indicator key assignments on page 293.

Table 80: Incoming Call Indicator key assignments

Key	Mnemonic	Meaning
00-19	CAx	Station Category Number (x = 1-7)
00-19	CFB	Call Forward Busy
00-19	CFN	Call Forward No Answer
00-19	DF0	Dial 0 fully restricted
00-19	DL0	Dial 0
00-19	IAT	Inter-attendant call
00-19	INT	Intercept
00-19	LCT	Lockout
00-19	LD0	Listed DN 0
00-19	LD1	Listed DN 1
00-19	LD2	Listed DN 2
00-19	LD3	Listed DN 3
00-19	MWC	Attendant Message Center
00-19	RLL	Recall
00-19	Rxxx	Route number

Vertical keystrip 3

This keystrip includes the following operating keys:

Release – Allows the attendant to release a call from the console. When the release lamp is lit, it indicates that no incoming calls are being presented to the console.

Loop key/lamps – Allows the attendant to answer and originate calls from the console. The first call in the attendant queue is automatically presented to an idle loop key. Subsequent calls are queued and presented to a loop key when the console becomes idle.

Three lamp indicators, positioned on the upper right-hand side of the keystrip, provide the following information:

- Two Alarm indicators : When steadily lit, the minor alarm lamp indicates the system has detected a malfunction that does not affect normal call processing. When the major alarm lamp is steadily lit, the system has detected a malfunction that does not permit normal call processing.

- Call Waiting indicator : The Call Waiting lamp indicates the number of calls in the attendant queue and the length of time they have been waiting to be answered. The lamp changes from steadily lit to flashing when waiting calls exceed a certain number, or when a call has been waiting longer than a specified time. The number of waiting calls are displayed by pressing the Display Calls Waiting key, if assigned.

Vertical keystrip 4

This keystrip provides the following fixed feature keys:

Hold – Allows the attendant to hold a call at the console.

Conference – Permits the attendant to set up a conference of up to five conferees, plus the attendant.

Release Destination – Allows the attendant to release the called party from a call held at the console, while holding the calling party.

Release Source – Allows the attendant to release the calling party from a call held at the console, while holding the called party.

Signal Source and Destination – Allows the attendant to recall either party to a call held on the console.

Exclude Destination – Excludes the called party from an established call held at the console, allowing the attendant to speak privately with the calling party.

Exclude Source – Excludes the calling party from an established call held at the console, allowing the attendant to speak privately with the called party.

Volume Control – Allows the attendant to change the volume of alerting signals. Each depression of the key changes the volume of the signal by one step in an eight step range.

Vertical keystrip 5

The optional features listed in Table 81: Attendant console optional feature key assignments on page 294 can be defined on this keystrip.

Table 81: Attendant console optional feature key assignments

Key	Mnemonic	Meaning
00	BVR	Busy Verify
01	BIN	Barge-In

Key	Mnemonic	Meaning
00-09	ADL	Autodial
02-09	AWU	Automatic Wake Up
00-09	CHG	Charge Account
00-09	CPN	Calling Party Number
00-09	DCW	Display Calls Waiting
00-09	DDL	Do-Not-Disturb, Individual
00-09	DDT	Display Date
00-09	DPD	Display Destination
00-09	DPS	Display Source
00-09	DTM	Display Time
02-09	EES	End-to-End Signaling
00-09	GND 0-99	Group Do-Not-Disturb
00-09	MCK	Message cancellation
00-09	MDT	Display/Change Date
00-09	MIK	Message indication
00-09	MTM	Display/Change Time
00-09	PAG xxx…x	Paging (xxx…x = route access code)
00-09	PRG	Attendant Administration
00-09	PRK	Call Park
00-09	RDL	Stored Number Redial
00-09	RTC	Routing Control
00-09	SCC xxxx	Speed Call Controller (xxxx = list number)
00-09	SSC xxxx	System Speed Call Controller (xxxx = list number)
00-09	TRC	Malicious Call Trace

The console has a Shift key on the fixed feature key strip that provides access to an Options menu. This menu allows the setting of the display screen contrast, buzz tone, language, time and date format, and calls waiting options. For more information about the Options menu, see *Telephones and Consoles Fundamentals* (NN43001-567).

The Shift key also allows the Avaya 2250 Attendant Console to have 20 Incoming Call Indicator keys in the regular mode, and 20 Trunk Group Busy keys and an additional ten flexible feature keys in the shift mode. Add-on modules are not required on the console to provide the additional key functions.

Attendant Call Party Name Display (CPND) and the Enhanced Busy Lamp Field/Console Graphics Module capabilities may be equipped with the Avaya 2250 Attendant Console. Refer to the feature modules in this document for a complete description of these capabilities.

For more information about attendant consoles and associated hardware, see:

- *Telephones and Consoles Fundamentals, NN43001-567*
- *Communication Server 1000M and Meridian 1 Large System Maintenance, NN43021-700*

Operating parameters

Refer to the preceding Avaya documents.

Feature interactions

Refer to the preceding Avaya documents.

Feature packaging

Attendant Console capabilities are included in base system software.

Calling Party Name Display (CPND) package 95 includes Attendant CPND and requires Digit Display (DDSP) package 19.

M2250 Attendant Console (DCON) package 140 requires M2000 Digital Sets (DSET) package 88.

Feature implementation

Task summary list

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

1. Table 82: LD 15 - Attendant console-related prompts and responses. on page 297
2. Table 83: LD 12 - Add an attendant console. on page 299

Table 82: LD 15 - Attendant console-related prompts and responses.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	LDN	Department Listed Directory Numbers.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- LDN0 - LDA0	xxx…x xx xx… ALL	Listed Directory Number 0. Attendant consoles associated with LDN0 (see Note).
- LDN1 - LDA1	xxx…x xx xx… ALL	Listed Directory Number 1. Attendant consoles associated with LDN1 (see Note).
- LDN2 - LDA2	xxx…x xx xx… ALL	Listed Directory Number 2. Attendant consoles associated with LDN2 (see Note).
- LDN3 - LDA3	xxx…x xx xx… ALL	Listed Directory Number 3. Attendant consoles associated with LDN3 (see Note).
TYPE:	NIT	Gate opener.
- NIT1	xxx…x	First Night Service DN.
- TIM1	hh mm	Hour and minute of first Night Service DN.
- NIT2	xxx…x	Second Night Service DN.
- TIM2	hh mm	Hour and minute for second Night Service DN.
- NIT3	xxx…x	Third Night Service DN.
- TIM3	hh mm	Hour and minute for third Night Service DN.
- NIT4	xxx…x	Fourth Night Service DN.
- TIM4	hh mm	Hour and minute for fourth Night Service DN.
TYPE:	ATT_DATA	Attendant console options.
ATDN	(0) xxx…x	Attendant DN.
- NCOS	(0)-99	Attendant Network Class of Service for all consoles.
TYPE:	CAS	Centralized Attendant Service options.
- CAS	(NO) YES	Change Centralized Attendant Service options.
TYPE:	ANI	Automatic Number Identification.
OPT	(IC1) IC2 (XTG) ITG	10 or 20 Incoming Call Indicators. Trunk Group Busy keys not equipped/equipped.

Prompt	Response	Description
	(LOD) LOA (XDP) IDP (XBL) IBL (SYD) SYA	(Deny) allow Lockout. Digit Display not equipped/equipped. Enhanced Busy Lamp Field not equipped/equipped. (Deny) allow Secrecy.
- ANAT	xxx x	Attendant Billing number.
- ANLD	xxx…x	ANI listed DN.
TYPE:	ATT_DATA	Attendant console options.
AATT	xxxx	AIOD attendant identifier.
TYPE:	TIM	Timers.
- RTIM	xxxx yyyy zzzz	Recall timers. xxxx = slow answer (0-378). yyyy = Camp-On (0-510). zzzz = Call Waiting (0-510).
- ATIM	(0)-126	Attendant Alternative Answering timer.
ICI	xx yyy	Incoming Call Indicator key assignment. xx = key number. yyy = mnemonic (see Table 80: Incoming Call Indicator key assignments on page 293). Multiple responses can be entered for the same key. To remove an entry, enter xx NUL, then reenter the desired responses. To add an entry, enter the desired response. It will be added to any already existing response.
- AQTT	0-(30)-255	Attendant queue timing threshold in seconds.
TYPE:	ATT_DATA	Attendant console options.
- AODN	xxxx…x	Attendant overflow DN.
TYPE:	PWD	Gate opener.
- ATAC	xxxx	Attendant Administration access code.
TYPE:	ATT_DATA	Attendant console options.
- CWUP	(NO), YES	Call Waiting queue update.
- CWCL	(0)-255 (0)-255	Call Waiting lower and upper thresholds for number of calls in queue.
- CWTM	(0)-511 (0)-511	Call Waiting lower and upper thresholds for time in queue.
- CWBZ	(NO) YES (NO) YES	Buzz when Call Waiting thresholds are exceeded. Buzz when first call enters queue.
- MATT	(NO) YES	Attendant consoles used as Message Center.
- SPVC	0-63	Attendant number for supervisor console.
TYPE:	AWU	Automatic Wake Up options.
- AWU	(NO) YES X	Enable Automatic Wake Up (X erases AWU information).

Prompt	Response	Description
- ATRC	(NO) YES	Attendant Recall after failed AWU attempts.
Enter one or more attendant numbers (1-63). Enter ALL to enable this listed DN on all attendants. Precede the attendant number with X to remove.		

Table 83: LD 12 - Add an attendant console.

Prompt	Response	Description
REQ	NEW	Add a console.
TYPE	2250 PWR	Attendant console type. Avaya 2250 Attendant Console. Power TN.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SETN		Second TN (must be on same loop as primary TN of attendant console).
	l s c u	Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
ANUM	1-63	Attendant number (1-63).
SSU	0-4095	System Speed Call user list number.
ICDR	(ICDD) ICDA	(Deny) allow internal call detail.
CPND	(CNDD) CNDA	(Deny) allow Call Party Name Display. Prompted if TYPE is 2250.
DNDI	(DNDD) DNDA	(Deny) allow dialed name display.
EBLF	(BLFD) BLFA	(Deny) allow enhanced busy lamp field. Prompted if TYPE is 2250.
AADN	xxx…x	Attendant Alternative Answering DN.
KEY	xx aaa	Key number and mnemonic for feature assignments (see Table 81: Attendant console optional feature key assignments on page 294).

Feature operation

Refer to the appropriate attendant console user guide for specific operation procedures.

Chapter 38: Attendant Delay

Contents

This section contains information on the following topics:

Feature description

The Attendant Delay feature prevents an attendant from performing the following operations during a customer defined period (0 to 14 seconds inclusive) after a call is presented or recalled to the attendant:

- placing the call on hold
- releasing the call
- parking the call
- extending the call
- performing call splitting
- activating paging
- placing a call, if Secrecy or Enhanced Secrecy applies to the presented call or recall

Operating parameters

If Night Service, Attendant Overflow Position, Position Busy, or Attendant Alternate Answering are active, calls presented or recalled to the attendant are automatically routed to a pre-selected station, and are not subject to Attendant Delay.

Feature interactions

Attendant Console Misoperation

Attendant Delay takes precedence over Attendant Console Misoperation.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

Table 84: LD 15 - Enable Attendant Delay.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data
TYPE:	TIM	Timers
- ADHT		Attendant Delay on Hold Timer
	(0)-14	Respond to the ADHT prompt with 0 (the default) to leave this feature disabled, or with a value from 1 to 14 seconds for the Attendant Delay timer to enable the feature. This must be done for each customer to be equipped with the feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 39: Attendant Display of Speed Call or Autodial

Contents

This section contains information on the following topics:

Feature description

With the Attendant Display of Speed Call or Autodial feature, when an attendant uses the Speed Call or Autodial feature to dial a number automatically, the dialed digits are shown on the console display. The speed-call code and the dialed speed-call number are displayed for a speed-call operation. The dialed autodial number is displayed for autodial operation.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

International Supplementary Features (SUPP) package 131.

Feature implementation

No specific implementation procedures are required to use this feature.

Feature operation

If an attendant presses the Speed Call key, the speed-call code and dialed speed call number are shown on the attendant console display.

If an attendant presses the Autodial key, the dialed autodial number is shown on the attendant console display.

Chapter 40: Attendant Forward No Answer

Contents

This section contains information on the following topics:

Feature description

The Attendant Forward No Answer (AFNA) feature is comprised of two capabilities. The first allows Direct Inward Dial (DID), Direct Outward Dial (DOD), or Central Office (CO) calls, presented to the attendant and not answered within a customer-defined period of time to be forwarded to another attendant, or, if the customer is in Night Service, to the night DN.

The second capability allows Direct Inward Dial or Central Office calls, presented to a station that is in Night Service, to be disconnected if not answered within the pre-defined ring cycle, or time period. This second capability is called Night Forward No Answer (NFNA).

Two timers are available: the Attendant Forward No Answer timer (AFNT) and the Attendant Forward No Answer Buzz timer (AFBT), both of which are programmed in LD 15.

If the AFBT timer is programmed, when a call is presented to the attendant, the attendant receives a buzz at maximum volume for the duration of the AFBT timer. If the value set for the AFNT timer is higher than that of the AFBT timer, the attendant receives a buzz at normal volume for the duration between when the two timers expire. The AFNT timer can be set between two and 126 seconds. The AFBT timer cannot be set higher than the AFNT timer.

If the attendant does not accept the call before the AFNT timer expires, the attendant is put in Position Busy and the call is relinked to the top of the queue. If all attendants are put in Position

Busy, the call can be forwarded using Attendant Overflow Position (AOP) or Night Service if equipped.

When a call is forwarded from the attendant queue to a busy Attendant Overflow Position, the call remains in the queue. If the AOP is idle, the Attendant Forward No Answer timer is started. If the call is not answered before time-out, the AOP is idled. The call is relinked to the top of the queue. If all attendants are in Position Busy, Night Service is activated and the call is transferred to the night DN.

If the night DN is busy, the call is added to the queue, provided the call involves a CO, FEX, WATS, CAS, or CAMA trunk, or was handled by Enhanced Night Service. Other calls, such as TIE or internal calls, are given busy tone.

During Night Service, when a DID or CO trunk call is presented to an idle DN, the Night Forward No Answer (NFNA) ring counter is started. If the call is not answered during the NFNA time cycle, the call is disconnected. Non-DID and non-CO calls ring until the call is answered or the calling party hangs up.

Operating parameters

Attendant Forward No Answer operates in a standalone or networking environment. For networking applications, the transferring and terminating stations can be located on different nodes.

Attendant Forward No Answer does not apply to inter-attendant calls.

Night Forward No Answer (NFNA) and Night Forward No Answer in seconds (NFNS) do not apply to calls waiting in the ACD queue or the Primary Line Directory Number (PLDN) queue.

When Night Forward No Answer times out on an unanswered trunk, the trunk is locked out until the far-end goes on-hook.

The maximum number of ring cycles for Attendant Forward No Answer on an Attendant Overflow Position is 63.

AFNA timing ceases and the volume of the attendant buzzer is set to the original value in the following cases:

- If the attendant answers a call
- If the attendant answers an Automatic Wake-up recall on the AWU key
- If an attendant-extended call is answered on a telephone during a slow answer recall to the attendant
- If a call waiting call is answered at a telephone while the attendant is ringing

If a telephone or trunk disconnects while the attendant is being rung, and the AFNA timing cannot continue on the source or destination side, the volume of the attendant buzzer is set to its original value.

The NFNS timing starts when a DID/DOD/CO call is recalled to the night station, as part of the Recall to Night Station treatment, requeued to the night station as part of the Requeueing of Attendant Presented Calls treatment, or rerouted to the night station as part of the Attendant Clearing During Night Service treatment.

If both the Disconnect Timer (DCTI) of the Periodic Clearing feature and NFNA or NFNS are defined, the first one which expires will disconnect a DID or CO call.

Feature interactions

AC15 Recall: Timed Reminder Recall

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

Attendant Recall

If an attendant recall is affected through the Attendant Recall key on a proprietary telephone, or through a switchhook flash on an analog (500/2500 type) telephone, the destination side on the console is not dropped before the call is routed to the night DN.

Camp-On to a telephone in Ringback or Dialing

Camp-on recall takes precedence over the Attendant Forward No Answer recall. However, if during the recall the customer goes into Night Service and the recall is not answered by the night DN, the call is disconnected according to the Attendant No Answer feature processing.

DPNSS1 Diversion

If an incoming call is handled for Network Attendant Services routing towards DPNSS1, no diversion signaling is sent back to the calling party.

Multi-party Operations - Recovery of Misoperation during Call Transfer

Multi-Party Operations – Recovery of Misoperation During Call Transfer takes precedence over NFNA and NFNS for DID/DOD/CO calls.

When a DID/DOD/CO call is transferred from one station to another station on the same node, Ring Again No Answer has priority over NFNA and NFNS.

Night Forward No Answer

Call Forward No Answer has priority over Night Forward No Answer and AFNA on the Attendant Overflow Position.

Night Service Enhancements

Any call which has been presented to the Attendant Overflow Position cannot be removed from the telephone and requeued by pressing the Make Set Busy (MSB) key. The call will only be removed if the Attendant Forward No Answer feature is active, and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Position Busy with Call on Hold

If an attendant with a call on hold does not answer an Attendant Forward No Answer call within a customer-defined time, the console is not placed in Position Busy.

Recall to Same Attendant

If the attendant does not answer a call and the Attendant Forward No Answer feature is equipped, the console is forced into the Position Busy state and the call is routed to the first available idle attendant.

Switchhook Flash

If a switchhook flash is performed on an analog (500/2500 type) telephone, the AFNA timing stops to allow for a valid disconnection. If a valid disconnection is not affected, the AFNA timing cycle begins again.

Feature packaging

Attendant Forward No Answer (AFNA) is package 134; however, this package is mutually exclusive with Attendant Alternate Answering (AAA) package 174.

Within a networking environment, Network Attendant Service (NAS) package 159 is required.

Feature implementation

Table 85: LD 15 - Modify data for each customer member to be configured.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	ATT_DATA	Attendant console options.
...		
- OPT	(DNCA) DNCS	If DNCA is entered, all DID/CO or DOD calls are disconnected after the number of ring cycles defined by the response to the NFNA prompt while the system is in Night Service. If DNCS is entered, outgoing CO/DOD calls or incoming CO/DID calls in the answered state, and waiting on a telephone are disconnected after the number of seconds defined in response to the NFNS prompt expires.
...		
- AFNT	(0)-2-126	Attendant Forward No Answer Timer. The number of seconds in two-second intervals that the call is presented to the attendant before Attendant Forward No Answer is attempted. Odd entries are rounded down to the next valid entry. If 0 is entered, the call is not forwarded.
- AFBT	(0)-2-x	Attendant Forward Buzz Tone, where:

Prompt	Response	Description
		x = the value defined for AFNT. The number of seconds in two-second intervals that the attendant is buzzed at full volume before the Attendant Forward No Answer timer is reached. Odd entries are rounded down to the next valid entry. If 0 is entered, the original volume is in effect.
TYPE:	TIM	Timers.
...		
- NFNA	(0)-63	Night Forward No Answer ring cycles (prompted if OPT = DNCA). The number of times a DID/DOD and CO trunk call will ring a telephone before being disconnected during Night Service. A default value of 0 causes the call not to be disconnected.
- NFNS	(0)-504	Night Forward No Answer in seconds (prompted if OPT = DNCS). If a value is entered for this prompt, all outgoing CO/DOD trunk calls in a waiting state, and all incoming CO/DID calls in the answered state will be disconnected after the time in seconds expires as entered in response to this prompt. The entered value must be a multiple of eight. A default value of 0 causes the call not to be disconnected.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 41: Attendant Incoming Call Indicators

Contents

This section contains information on the following topics:

Feature description

Attendant consoles can be equipped with up to 20 Incoming Call Indicator (ICI) key/lamp pairs to identify the type of calls being presented and the call status for each particular call type. The customer can specify which incoming call types are to be assigned a separate ICI key. Possible call types include, but are not limited to, the following:

- Trunk calls (such as FX, WATS, and TIE)
- Listed Directory Number (LDN) calls
- Dial zero calls
- Fully restricted dial zero calls
- Automatic Timed Reminder recalls
- Attendant Interpositional calls
- Attendant Intercept calls

- Call Forward Busy calls
- Call Forward No Answer calls

Three lamp states are associated with each Incoming Call Indicator key (See Table 86: Key lamp states associated with each Incoming Call Indicator key on page 314).

Table 86: Key lamp states associated with each Incoming Call Indicator key

Lamp state	Status of call type
Dark	No calls of this type are waiting.
Flashing	One call of this type is waiting in queue.
Steadily lit	Two or more calls of this type are queued, or one call has been waiting longer than 20 seconds.

Operating parameters

The ICI feature applies to attendant consoles only.

The number of ICI keys to be assigned (10 or 20) is defined in the Customer Data block. The default is ten.

No more than 20 ICI key/lamp pairs can be assigned to an attendant console. The assignment of call types to ICI key/lamp pairs is flexible. All attendant consoles in the customer group will have the same ICI key assignments.

Feature interactions

Attendant Call Selection Call Waiting

The ICI feature is used with the Attendant Call Selection and Call Waiting features to recognize, answer, and process incoming calls.

DPNSS1 Night Service

When a Night Service call is diverted to an attendant, the Incoming Call Indicator is the number of the incoming route. This is the same as for a NAS MCDN call routed to an attendant.

ISDN Semi Permanent Connections for Australia

Calls using an ISPC link are always presented as calls over TIE trunks.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 87: LD 15 - Assign ICI keys for attendant consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- OPT	(IC1) IC2	10 or 20 Incoming Call Indicators.
- ICI	0-19 CAx 0-19 CFB 0-19 CFN 0-19 DF0 0-19 DL0 0-19 IAT 0-19 INT 0-19 LCT 0-19 LD0-3 0-19 MWC 0-19 RLL 0-19 xxx	Station category number. x = category number 1 through 7. Call Forward Busy. Call Forward No Answer. Dial 0 fully restricted. Dial 0 (attendant). Inter-attendant call. Call intercept. Line Lockout Intercept. Listed Directory Number (0 through 3). Attendant Message Center. Recall. Route number.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 42: Attendant Interpositional Transfer

Contents

This section contains information on the following topics:

Feature description

Attendant Interpositional Transfer enables an attendant to call or transfer a call to another attendant in a multiple console group, even when the destination attendant console is busy.

When transferring a call to another attendant whose console is idle, the interpositional call is presented immediately. If the called attendant is busy, the calling attendant hears a busy tone. The attendant then presses the Release key and the transferred call will be the next call presented to the called attendant console.

Operating parameters

A call can be transferred to an attendant console in the Position Busy state; however, the called console does not receive any audible signal. A Call Waiting indication appears on the console display.

Feature interactions

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

The Attendant Overflow Position feature is supported in a UDP DPNSS1 network. An attendant can call or transfer a call to another attendant in a multiple-console group, even when the destination attendant console is busy.

Network Attendant Service

An attendant is not able to call a specific attendant on another node by dialing the attendant DN followed by the attendant number. The attendant dials the NARS or CDP or LDN number the same as a telephone dials to reach the attendants at another node.

Night Service Enhancements

The requeuing of interpositional calls is not allowed. Night Service enhancements do not apply to interpositional calls, which remain on the console until answered.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 88: LD 15 - Add/change an Interpositional Call Incoming Call Indicator (ICI) key on attendant consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.

Prompt	Response	Description
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- ICI	0-19 IAT	Add an Inter-attendant Call ICI to all consoles.

Feature operation

To transfer a call to a busy attendant (attendant console):

- Press **Rls**. Your call will be the next call presented to the busy attendant.

To transfer a call to an attendant console in Position Busy mode:

- Dial the Interpositional access code (0) and the desired attendant position number. You receive a busy tone. Press **Rls**.

To answer a call transferred to an attendant console in Position Busy mode, follow these steps:

1. The Call Waiting indicator lights; there are no audible tones. Press the **Position Busy** key to take the console out of Position Busy mode.
2. The call is presented to the Loop key and you receive an audible tone. Press the **Loop** key.

Chapter 43: Attendant Lockout

Contents

This section contains information on the following topics:

Feature description

Attendant Lockout restricts the attendant from entering an established connection completed through and held on the console. Attendant Lockout does not come into effect until the call has been answered.

The attendant can re-enter the call if the source party is a station telephone. Attendant Lockout occurs only if the source party is an external number (trunk), and the destination party is a telephone.

Operating parameters

Busy Verify and Barge-In allow the attendant to override the Attendant Lockout feature.

Feature interactions

Attendant Recall

If one of the stations activates Attendant Recall, the attendant is allowed to re-enter the connection.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 89: LD 15 - Allow/deny Lockout for attendant consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- OPT	(LOD) LOA	(Deny) allow attendant lockout.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 44: Attendant Overflow Position

Contents

This section contains information on the following topics:

Feature description

Attendant Overflow Position (AOP) allows certain types of calls to be automatically rerouted to a specified idle Directory Number (AOP DN) when calls waiting to be answered have exceeded a defined threshold, or an attendant is in the Position Busy state, but the system is not in Night Service.

When a call that can be rerouted has been waiting longer than the customer-defined Attendant Queue Timing Threshold (0-255 seconds), it is rerouted to the AOP DN. Calls that can be rerouted to the AOP DN are trunk calls, internal calls and Call Forward Busy, or Call Forward No Answer calls directed to the attendant.

Attendant calls that cannot be rerouted are transfer calls, intercept calls, parked call recalls, automatic or manual recalls, and attendant interposition calls. These calls will not be answered until an attendant becomes available.

When the last attendant console is put into Position Busy or disabled, the system does not go into Night Service if an AOP DN is available. In this case, calls that can be rerouted will be forwarded to the AOP DN. Ineligible calls remain unanswered until the system is put in Night Service or one of the consoles deactivates Position Busy.

Operating parameters

An AOP DN can be a single-appearance, multiple-appearance single-call, or multiple-appearance multiple-call DN. If it is a Multiple Appearance DN, a proprietary telephone can busy out the AOP DN for all appearances.

An analog (500/2500 type) telephone can have an AOP DN. It does not have the ability to busy out the AOP DN and continue to receive calls. If it is a requirement that the analog (500/2500 type) telephone have an AOP DN, the AOP DN must also appear on a proprietary telephone to create a mix of telephones, which negates privacy.

In order to properly identify and greet attendant overflow calls, it is best to have the AOP DN appear on a proprietary telephone's secondary DN.

Proprietary telephones specified as Attendant Overflow Positions can prevent calls from being rerouted by the Attendant Overflow feature. To prevent attendant overflow calls, press the Attendant Overflow Position Busy (AOP Busy) key/lamp pair on the telephone. Activating this key will busy out all appearances of the AOP for either Single Call Ringing or Multiple Call Ringing arrangements. Overflow calls will remain in the attendant queue. Normal incoming calls to the AOP telephone will not be affected.

The following requirements apply to the activation/deactivation of the AOP Busy key:

- A telephone with an AOP Busy key must have an appearance of the AOP DN in order for the key to work.
- Any AOP DN that has an AOP Busy key can activate or deactivate the AOP feature. If the AOP Busy key is activated at one appearance of the AOP DN, attendant calls are not rerouted to any appearance of the AOP DN.
- Activation or deactivation of the AOP Busy key does not affect any call already rerouted to the AOP DN.
- If all consoles are in Position Busy and the system is not in Night Service when an AOP Busy key is activated, the system goes into Night Service.
- If the system is in Night Service when the AOP Busy key is deactivated, the system remains in Night Service.
- Activation or deactivation of the AOP Busy key does not affect the Position Busy status of the attendant console. If all attendant consoles are in Position Busy and the AOP Busy key is activated, the system goes into Night Service.
- The status of the AOP Busy key remains unchanged through a system initialization but is deactivated if a system reload occurs.

The CAS to AOP Interworking feature allows both Centralized Attendant Service-Main (CASM), or Centralized Attendant Service-Remote (CASR), and Attendant Overflow Position packages to be configured and co-exist in a network. In an environment where both packages are configured, CAS takes precedence over AOP.

Each customer may have only one AOP DN. The AOP DN cannot be a private line DN, a trunk DN, a Control DN, a BRI DN, or a SPRE code.

There are no special ringing cadences or lamp operations to indicate that an incoming call to the AOP DN is an Attendant Overflow Position call. It is recommended that the AOP DN be used only for Attendant Overflow Position calls enabling calls to be answered appropriately.

If the AOP DN is busy, calls remain in the attendant queue and are not rerouted through the Attendant Overflow Position feature until the DN is free to receive the next call.

Calls will not be rerouted to the Attendant Overflow Position DN when

- Calls are on an Integrated Services Digital Network (ISDN) or Electronic Switched Network (ESN) network.
- All appearances of the AOP DN are busy.
- The AOP DN is in the Call Forward All Calls mode.
- The call is an interposition call from an attendant.
- The call has been redirected to the attendant by the Call Transfer or Attendant Recall features.
- The call is an intercept call to the attendants.
- The system is in the Power Fail Transfer mode.
- All appearances of the AOP DN have the Make Set Busy feature activated.
- Any appearance of the AOP DN has activated Attendant Overflow Position Busy (AOP Busy).
- An analog (500/2500 type) telephone appearance of the AOP DN goes idle and a Call Waiting call is queued for the telephone. The Call Waiting call rings the telephone and AOP calls are not rerouted to the telephone.
- The AOP DN goes idle with a Camp-On call queued for the telephone. The Camp-On call rings the telephone and AOP calls are not rerouted to the telephone.
- The rerouting of the call violates the access restrictions or Class of Service restrictions on the AOP DN telephone. For example, if the AOP DN is FR2, an external Public Exchange network call will not be rerouted to the AOP DN because it is prohibited by the telephone access restrictions.
- The system is in Night Service.

Feature interactions

AC15 Recall: Timed Reminder Recall

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

Attendant

The Calls Waiting indicator on the attendant console is updated when a call is rerouted to the AOP DN.

Attendant Overflow Position Busy

If the telephone with Attendant Overflow Position (AOP) DN has an Attendant Overflow Position Busy (AOP Busy) key activated, calls will not overflow to any appearance of the AOP DN.

Attendant Recall

An Attendant Overflow Position call answered at an AOP DN may be recalled to the attendant using the Attendant Recall capability (ARC key).

Attendant Timed Recall Automatic Timed Reminders

After an attendant call has been rerouted using the AOP feature, there is no automatic timed recall to the attendant or any other DN.

Automatic Call Distribution

Externally marked trunks will overflow to an Automatic Call Distribution (ACD) DN. The ACD DN may only be an ACD agent configured as a virtual Voice Mail System agent (for example, Avaya CallPilot).

Automatic Wake Up

Automatic Wake Up recalls are not redirected to a customer-defined Attendant Overflow Position DN. Failed wake up calls stay in the attendant queue or ring indefinitely on the console.

Call Forward All Calls

If the telephone assigned an Attendant Overflow DN has activated the Call Forward All Calls feature, overflow calls are not rerouted to the telephone. If an analog (500/2500 type) telephone is forwarded, AOP is canceled.

Call Forward, Internal Calls

If Attendant Overflow redirects an internal call to a telephone that is Internal Call Forward active, the call will remain in the attendant queue, and will not receive Internal CFW treatment.

Call Forward No Answer

A call rerouted through Attendant Overflow Position will Call Forward to the forwarding DN only if it is the Prime DN or a single appearance DN on that telephone.

Call Pickup

An Attendant Overflow Position Call presented to the AOP DN can be picked up by any station belonging to the same Call Pickup Group.

Conference

An Attendant Overflow Position call answered on an AOP DN may be conferenced with another DN.

Departmental Listed Directory Number

Listed Directory Number calls that have been waiting in the queue longer than the specified threshold period will be routed to the Attendant Overflow Position.

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

The Attendant Overflow Position feature is supported on a UDP DPNSS1 network. If an incoming DPNSS1 UDP call is queued to the attendant, and if the call is not answered within a predefined period of time, the call can be redirected to the Attendant Overflow DN.

Flexible Attendant Call Waiting Thresholds

The Attendant Overflow Position is not counted as an active attendant.

Flexible Line Lockout

A call intercepted to the attendant due to Flexible Line Lockout receives Attendant Overflow Position (AOP) treatment if the feature package is equipped and the AOP Directory Number (DN) is defined.

Group Hunt

A PLDN cannot be configured as an Attendant Overflow DN (AODN).

Line Lockout

If a telephone with an AOP DN is in Line Lockout, it still receives AOP calls.

Make Set Busy

If a telephone that is the only idle AOP DN has MSB activated, calls will not overflow.

If the AOP DN is a multiple appearance DN, the MSB key should be added to all telephones with an AOP DN.

If MSB is activated in a Multiple Call Ringing arrangement, the telephone appears busy. All other appearances of the AOP DN will still receive calls. This allows the user to leave the telephone and prevent callers from overflowing and receiving ringback with no answer.

If the AOP DN is a Multiple Appearance, Single Call arrangement and MSB is activated, the AOP DN of that telephone will flash, but the telephone will not ring (the call can still be answered from that appearance).

Manual Line Service

When Attendant Overflow Position (AOP) is defined, Manual Line Service follows the AOP directions.

Meridian Hospitality Voice Services

Attendant Overflow Position (AOP) allows unanswered calls to the attendant to be forwarded to a customer-defined Directory Number (DN) after a defined time. A call can also be overflowed if all the attendants are in Position Busy State. With AOP equipped, overflowed calls can be directed to Avaya CallPilot. The AOP DN must be defined as an Automatic Call Distribution (ACD) Directory Number (DN), and the ACD DN must have an ACD agent assigned as a virtual VMS agent.

Multiple Appearance Directory Number

A multiple appearance, multiple call AOP DN allows as many overflow calls to be in progress as there are appearances of the DN. A multiple appearance, single call AOP DN allows only one overflow call at a time.

Night Key for Direct Inward Dialing Digit Manipulation

When the last attendant activates the POS BUSY key, the system does not go into Night Service if an Attendant Overflow Position Directory Number (DN) is available.

Night Service

A call rerouted through the Attendant Overflow Position feature is not redirected to the Night DN if the system is subsequently put into Night Service. When all attendant consoles are in Position Busy, the system will not go into Night Service until the AOP Busy key is activated.

Deactivating the AOP Busy key after the system has been placed in Night Service does not affect the Night Service feature.

Night Service Enhancements

If a call with a ringing party on the destination side is presented at the last-active attendant console, and there is an active Attendant Overflow Position, the ringing destination will be disconnected when the call is requeued. Likewise, if the call is a Call Waiting recall, Call Waiting will be canceled.

Night Service Enhancements/Network Attendant Service (NAS)

The routing configuration for NAS will apply during Night Service. External calls and recalls may be queued to a remote Night DN, if defined. Internal calls and internal recalls queued during Day Service will be dropped, if the Night DN has been defined on a remote node.

Recall to Same Attendant

Recalls and inter-attendant calls are not routed to the Attendant Overflow Position.

Ring Again

If Ring Again is activated against the AOP DN, notification is given to the originator when the telephone becomes idle. An AOP call, however, takes precedence over Ring Again notification on the AOP DN when the AOP DN becomes free.

Traffic Measurement

Traffic measurements are provided for the Attendant Overflow feature in Traffic Report TFC005. A count of the number of attendant calls rerouted through the feature is printed.

Feature packaging

Attendant Overflow Position (AOP) package 56 has no feature package dependencies. Attendant Overflow Position and Centralized Attendant Service are, however, mutually exclusive.

Feature implementation

Task summary list

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

1. Table 90: LD 15 - Assign/change an Attendant Overflow Position DN and queue threshold timing. on page 331
2. Table 91: LD 11 - Add/change an AOP DN and AOP Busy key. on page 331
3. Table 92: LD 10 - Add/change an Attendant Overflow Position DN on an analog (500/2500 type) telephone. on page 332

Table 90: LD 15 - Assign/change an Attendant Overflow Position DN and queue threshold timing.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- AQTT	0-(30)-255	Attendant queue timing threshold (AQTT).
- AODN	xxx...x	DN where calls are to be overflowed when they have been in queue the time specified for AQTT.

Table 91: LD 11 - Add/change an AOP DN and AOP Busy key.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx yyy...y	Attendant Overflow Position DN. xx = key number. yyy...y = DN.
KEY	xx OVB	Attendant Overflow Position Busy key.

Table 92: LD 10 - Add/change an Attendant Overflow Position DN on an analog (500/2500 type) telephone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DN	yyy...y	Attendant Overflow Position DN.

Feature operation

Attendant Overflow Position calls will be rerouted to all appearances of the AOP DN as long as the following conditions are met:

- The system is not in Night Service.
- The Attendant Overflow key (any AOP DN appearance) is not activated.
- At least one appearance of the AOP DN is on a telephone that does not have Make Set Busy activated.

To prevent attendant overflow calls from being rerouted to the AOP DN, do any of the following:

- Activate the Attendant Overflow Position Busy key.
- Activate the Make Set Busy key on all telephones with an appearance of the AOP DN.
- Place the system in Night Service.

To prevent attendant overflow calls from being rerouted to a single telephone with an appearance of the AOP DN (but not others):

- Activate Make Set Busy or
- Activate Call Forward All Calls (analog (500/2500 type) telephone).

Chapter 45: Attendant Position Busy

Contents

This section contains information on the following topics:

Feature description

If multiple consoles are defined for a customer, an attendant can remove a console from service by pressing the Position Busy key. Incoming calls are then directed to other consoles in the customer group.

Operating parameters

Position Busy applies to attendant consoles only.

Feature interactions

Attendant Administration

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

Attendant Supervisory Console

Activation of the Position Busy key on a Supervisory console puts the console in the supervisory mode.

Departmental Listed Directory Number

If all attendant consoles in an LDN group are in a Position Busy state, calls to that LDN will not be automatically presented to any attendant console in the customer group. Other attendants may only answer those LDN calls if the LDN has been assigned to an ICI key.

End-to-End Signaling

Attendant Position Busy works together with Attendant End-to-End Signaling (AEES). However, do not press this feature key while using AEES, or the Dual-tone Multifrequency (DTMF) code signals may be blocked.

Night Service

When the last console operator activates the Position Busy key or the Night key, Night Service is put into effect. Incoming calls receive the customer-specified night treatment.

When all attendants activate the Position Busy key, Night Service is in effect unless the Attendant Overflow Position (AOP) feature is equipped. If AOP is equipped, the Night key must be pressed to invoke Night Service. A call that is rerouted due to AOP is not redirected to the Night DN if the system is subsequently put into Night Service.

Night Service Enhancements

Any call that has been presented to the Attendant Overflow Position cannot be removed from the console and requeued by pressing the Make Set Busy (MSB) key. The call will be removed only if the Attendant Forward No Answer feature is active and the Attendant Forward No Answer Timer has timed out. In this case, the call is requeued and the Attendant Overflow Position is idled.

Recall to Same Attendant

If an attendant console is in maintenance or Position Busy when a Recall to Same Attendant call is recalled to it, the recall is presented to the first available idle attendant. If an attendant goes into Position Busy with a Return to Same Attendant call in Call Waiting, the waiting call is presented to the first available attendant.

Series Call

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

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

In a multi-console environment, press the Position Busy key on an attendant console to remove it from service.

Chapter 46: Attendant Recall

Contents

This section contains information on the following topics:

Feature description

Attendant Recall allows a user to call the attendant directly during an established call by pressing a single key. A three-way connection is established among the user, the attendant, and the third party.

To activate this feature, a separate Attendant Recall key/lamp pair must be equipped on proprietary telephones.

On single-line telephones, a user can recall the attendant during an established call by flashing the switchhook. Attendant Recall is automatic if a Transfer Denied Class of Service (XFD) is specified for the telephone. If a Transfer Allowed Class of Service (XFA) is specified, the user hears a special dial tone following the switchhook flash, and then dials zero (0) to recall the attendant. After a switchhook flash has been used to recall the attendant, it is not possible to return to a two-party connection before the attendant answers.

Operating parameters

In order for the Overflow Position Busy (OVB) key to work, the telephone must have an AOP DN configured.

Feature interactions

Attendant Alternative Answering

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

Attendant Forward No Answer

If an attendant recall is affected through the Attendant Recall key on a proprietary telephone, or through a switchhook flash on an analog (500/2500 type) telephone, the destination side on the console is not dropped before the call is routed to the night DN.

Attendant Lockout

If one of the stations activates Attendant Recall, the attendant is allowed to reenter the connection.

Attendant Overflow Position

An Attendant Overflow Position call answered at an AOP DN may be recalled to the attendant using the Attendant Recall capability (ARC key).

Attendant Secrecy

Attendant Secrecy does not apply on an attendant recall or when the attendant reenters a call held on a Loop key. The Exclude Source and Destination keys are used in these cases.

Attendant Splitting

After the attendant and the two parties have been connected, the attendant can use the Attendant Splitting feature to communicate separately with either party.

Automatic Redial

When an Automatic Redial (ARDL) call is not accepted by the calling party, the Attendant Recall (ARC) key is ignored.

Call Party Name Display

Attendant Recall using the Attendant Recall key or a switchhook flash results in both source and destination information being displayed. No redirection reason is displayed, however. In this type of recall, the party that pressed the Attendant Recall key or switchhook is the destination party.

Attendant Recall using Call Transfer or Conference displays the recalling party's DN and CPND information on the attendant's source line. No redirection reason is displayed. If the recall is done with the Transfer key the third party's DN and CPND information are displayed on the source line when the transfer is complete.

Directory Number Delayed Ringing

If a dialed telephone has Directory Number Delayed Ringing (DNDR) defined, and an attendant re-extends a call without releasing it, the DNDR timing is not reset. If the value of the recall timer is less than that of the DNDR timer, the call is recalled to the attendant before audible notification begins.

Direct Inward Dialing Call Forward No Answer Timer

The Direct Inward Dialing Call Forward No Answer Timer does not apply to an answered DID call that is extended to an unanswered station by the attendant – the call is recalled to the attendant using the Attendant Recall feature.

In-Band Automatic Number Identification

If an Automatic Call Distribution Agent is active on an IANI call and activates the Attendant Recall (ARC) key to call the attendant, the agent's display shows the attendant number when the attendant answers the call. The ANI number reappears when the attendant releases.

Incoming Call Indicator Enhancement

If an RDI-intercepted call that is extended by the attendant to the destination party having RDI Class of Service is either transferred back or recalled to the attendant, then the attendant recall ICI lights up and not the RDI-intercept ICI.

ICP Network Screen Activation and Flexible DN

When a call from another node is recalled to the Intercept Computer (ICP) position attendant, it is presented on the ICP terminal.

Multi-Party Operations

Users of analog (500/2500 type) telephones can perform an attendant recall during a two-party connection by performing a switchhook flash and then dialing the attendant DN.

Ring Again on No Answer

A telephone that is recalling the attendant cannot apply Ring Again on No Answer.

Secrecy Enhancement

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

Slow Answer Recall for Transferred External Trunks

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

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 93: LD 15 - Add/change a Recall Incoming Call Indicator (ICI) key on attendant consoles. on page 341
2. Table 94: LD 10 - Implement Attendant Recall for analog (500/2500 type) telephones. on page 342
3. Table 95: LD 11 - Add/change an Attendant Recall key for proprietary telephones. on page 342

Table 93: LD 15 - Add/change a Recall Incoming Call Indicator (ICI) key on attendant consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.

Prompt	Response	Description
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- ICI	xx RLL	Add a Recall ICI to all consoles.

Table 94: LD 10 - Implement Attendant Recall for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(XFD), XFA	(Deny) allow call transfer, which allows automatic Attendant Recall.

Table 95: LD 11 - Add/change an Attendant Recall key for proprietary telephones.

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

Feature operation

To contact an attendant during a call (Meridian 1 proprietary telephone), follow these steps:

1. Press **Att Recall**.
2. Stay on the line until the attendant answers.
3. When you hang up, the other party remains connected to the attendant.

To contact an attendant during a call (analog (500/2500 type) telephone with Transfer Allowed Class of Service), follow these steps:

1. Flash the switchhook (you hear a special dial tone).
2. Dial zero (0).
3. When you hang up, the other party remains connected to the attendant.

To contact an attendant during a call (analog (500/2500 type) telephone with Transfer Denied Class of Service), follow these steps:

1. Flash the switchhook (the attendant is automatically dialed).
2. When you hang up, the other party remains connected to the attendant.

Chapter 47: Attendant Recall with Splitting

Contents

This section contains information on the following topics:

Feature description

The Attendant Recall with Splitting feature provides an enhancement to the operation of the attendant console with the following features:

- Attendant Recall
- Call Transfer

This feature allows calls transferred to the attendant by the above features to be presented on the console loop with both the transferring and transferred parties on the console loop, with the transferred party automatically excluded if OPT in LD 15 is set to either SYA (Secrecy Allowed) or EHS (Enhanced Secrecy). Upon answering the call, the attendant then assumes control over both the transferred and transferring parties. The operation will also allow the transferring party to have control over the call as long as the call has not been answered by the attendant (example, the transferring party will be able to cancel the call transfer and return to the transferred party).

It is important to note that this enhancement applies to calls transferred to the attendant using Attendant Recall and Call Transfer only. Calls transferred to the attendant through operation of the Conference key on proprietary telephones, or through the operation of the Interpositional Call Transfers, do not receive splitting.

Operating parameters

This feature applies only to calls which arrive at the attendant by way of Attendant Recall or Call Transfer.

This feature will not function across a network.

This feature requires OPT in LD 15 (Customer Data Block) be set to either SYA or EHS.

Feature interactions

Attendant Secrecy

Secrecy Allowed (SYA)

If Secrecy is allowed at the attendant console, a two-party connection will be made only when the attendant answers the call. The attendant can converse privately with either the source or the destination side (Splitting) until the Loop key is pressed and a three-party connection is reestablished.

Secrecy Denied (SYD)

If Secrecy is denied at the attendant console, a three-way connection will be established between the transferring party, transferred party, and the attendant when the attendant answers the call.

Enhanced Secrecy (EHS)

Same as Secrecy Allowed except that a warning tone is included as part of all conversations involving the attendant and two or more parties to indicate that privacy has been interrupted.

Automatic Call Distribution (ACD)

A recall from an ACD DN to the attendant console will also activate the Attendant Recall with Splitting feature. The call is treated as if it had come from a normal internal DN instead of an ACD agent. The operation is described in Normal Operation on page 349

Automatic Hold

This feature does not have precedence over Attendant Recall (that is, automatic hold cannot be activated until the attendant answers the recall presented on the console). However, it can be activated even before the attendant answers a call transferred to the console.

Call Detail Recording (CDR) on Multiple Call Transfer

With PPM

Whenever a PPM call is transferred, the pulses accumulated against the current station that is responsible for this segment of the call are added to its terminal meter and a CDR X (an S for the first time) record is printed. When the call is eventually terminated, a CDR E record is printed.

Without PPM

The type and number of CDR records printed will be the same as the case for outgoing PPM call. The only difference is that no accumulated pulses will be included as part of the CDR messages.

Intercept Computer Dial from Directory

If a telephone transfers a call to the attendant, or a Meridian 1 proprietary telephone presses the Attendant Recall (ARC) key and the transferring party has not yet completed the transfer before the attendant answers, it is not possible to dial from the Intercept Computer (since the transferred party is connected to SRC, and the transferring party is connected to DEST).

Call Party Name Display

For the Avaya 2250 Attendant Console, M2317, and Meridian Modular Telephones, the appropriate DN and calling party's name will be correctly shown on the digit display when the attendant presses either the Exclude Source or the Exclude Destination key.

Multi-Party Operations

The Multi-Party Operations (MPO) feature introduces a new Class of Service; Three Parties Service Allowed (TSA), for analog (500/2500 type) telephones. It allows certain keys on these telephones to be programmed for conference, toggle between telephones, and disconnect. However, the toggle function will be disabled if a call is transferred to the attendant because of the Attendant Recall with Splitting feature.

Slow Answer Recall Enhancement

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

Transfer Restricted

This feature ignores the use of switchhook flash on analog (500/2500 type) telephones and as a result call transfer, conference, and attendant recall (with or without splitting) will not be allowed on a telephone basis.

Feature packaging

Attendant Recall with Splitting requires International Supplementary Features (SUPP) package 131.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

Normal Operation

The following events describe the normal operation whenever a call is transferred to the attendant using the Call Transfer feature at an analog (500/2500 type) telephone, or using the Call Transfer/Attendant Recall feature at a proprietary telephone, or using the operation of a register recall at an analog (500/2500 type) telephone with Transfer Denied Class of Service.

If an attendant console is idle, then the call will be presented to the console as follows:

- The Loop indicator stays off.
- The Recall Incoming Call Identification (ICI) indicator is turned on (with other ICI indicators associated with waiting calls of other types).
- The Source indicator for the loop on which the call is presented is turned on.
- The Destination indicator for the same loop flashes at 120 ipm.
- The console buzzes.
- The attendant console digit display indicates the DN (or name if CPND package is enabled) of the transferring party.
- The transferring party receives ringback tone.
- The transferred party is put on hold.

If no attendant console is idle when the transferring party dials the attendant access code, then the call is placed in the attendant queue, and the transferring party receives ringback tone. When this call moves to the top of the queue and an attendant console becomes idle, then the call will be presented to the console as described in the previous paragraph.

The attendant can then answer this call by pressing the Loop key, or by pressing the Recall ICI key. When the call is answered, the following occurs:

- The Loop indicator is turned on.
- The Recall ICI indicator stays on, and all other ICI indicators are turned off.
- The controlling party is presented at the attendant console as a destination, and the Destination indicator stays on steadily.
- The transferred party is presented at the console as a source, and the Source indicator remains on.

- The source (the transferred party) is automatically excluded from the connection, and the Exclude Source indicator is turned on.
- The destination (the transferring party) is connected to the attendant.

  The previous two events only occur when the SYA or EHS option is allowed. If SYD is defined in the Customer Data Block, a three party conference will be set up instead.
- If the call is transferred from a proprietary telephone, the proprietary telephone's Attendant Recall indicator or the Call Transfer indicator is turned off.

If the call is transferred from an M2317 telephone, then the screen on the corresponding telephone will go to the established state.

The attendant then assumes control over both the source and the destination as if both parties have been dialed by the attendant.

However, the transferring party can either return to the transferred party or complete the transfer operation any time before the attendant answers the call (that is, while the call is presented to the console, or placed in the attendant queue).

Cancel Call Transfer

The station user can return to the original party (the transferred party), before the attendant answers the call, as follows:

- The analog (500/2500 type) telephone user: By operating the register recall again, which causes the call to revert back to a two-party call, and the call to the attendant to be canceled.
- The proprietary telephone user: By pressing the DN key (DN indicator flashes at 120 ipm), which causes the call to revert back to a two-party call, the call to the attendant to be canceled, the DN indicator to stop flashing and stay on steadily, and the Call Transfer indicator (or the attendant Recall indicator) to turn off.
- By pressing the Release key or going on-hook, which causes the call to revert back to a two-party call and to be put on hold, the call to the attendant to be canceled, the Call Transfer indicator (or the attendant Recall indicator) to turn off, and the DN indicator to flash at 120 ipm.
- The M2317 telephone user: By pressing the DN key (DN indicator flashes at 120 ipm), which causes the call to revert back to a two-party call, the call to the attendant to be canceled, the DN indicator to turn on steadily.

  By pressing the Release key or going on-hook, which causes the call to revert back to a two-party call and to be put on hold, the call to the attendant to be canceled, (the attendant Recall indicator to turn off), and the DN indicator to flash at 120 ipm.

Pressing the DN key or operating the recall after the attendant answers the recall will be ignored.

Complete Call Transfer

While waiting for the attendant to answer the recall (ringback tone is received), the station user can complete the call transfer to the attendant as follows:

- The analog (500/2500 type) telephone user: By going on-hook, which causes the analog (500/2500 type) telephone to become idle and the attendant will ring.
- The proprietary telephone user: By pressing the Call Transfer key (or the attendant Recall key), which causes the DN indicator to turn off, the Call Transfer indicator (or the attendant Recall indicator) to turn off, and the DN to become idle.
- The M2317 telephone user: By pressing the CONNECT soft key (or the attendant Recall key), which causes the DN indicator to turn off, (the attendant Recall indicator to turn off), and the DN to become idle.

If the transfer operation is completed while the call is presented to the console, then the following will occur:

- The Destination indicator turns off.
- The Source indicator stays on steadily.
- The attendant console digit display changes to identify the transferred party.
- The transferred party receives ringback tone.
- The Recall ICI indicator stays on steadily (with other ICI indicators associated with waiting calls of other types).
- The console continues to buzz.

If the transfer operation is completed while the recall is in the attendant queue, then the DN at which the call is transferred becomes idle, the transferred party receives ringback tone, and the call stays in the queue as a recall.

Operation is not allowed after the attendant answers the recall. The transferring party cannot drop from the call in this case until the attendant presses the Release Destination key.

Chapter 48: Attendant Secrecy

Contents

This section contains information on the following topics:

Feature description

Attendant Secrecy automatically prevents a voice connection between the source and destination parties of a call being extended by an attendant, until the attendant connects the two parties. This allows the attendant to converse privately with the destination party before completing the connection. Attendant Secrecy is allowed or denied on a customer basis.

Operating parameters

Attendant Secrecy is available on attendant consoles only.

Attendant Secrecy operates only on external calls received from an outside trunk (for example, Central Office or WATS trunks).

Attendant Secrecy is not applicable to Integrated Services Access (ISA) trunks.

Feature interactions

AC15 Recall: Timed Reminder Recall

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

Attendant Recall

Attendant Secrecy does not apply on an attendant recall or when the attendant reenters a call held on a Loop key. The Exclude Source and Destination keys are used in these cases.

Attendant Recall with Splitting

Secrecy Allowed (SYA)

If Secrecy is allowed at the attendant console, a two-party connection will be made only when the attendant answers the call. The attendant can converse privately with either the source or the destination side (Splitting) until the Loop key is pressed and a three-party connection is reestablished.

Secrecy Denied (SYD)

If Secrecy is denied at the attendant console, a three-way connection will be established between the transferring party, transferred party, and the attendant when the attendant answers the call.

Enhanced Secrecy (EHS)

Same as Secrecy Allowed except that a warning tone is included as part of all conversations involving the attendant and two or more parties to indicate that privacy has been interrupted.

Console Presentation Group Level Services

The Secrecy option specified for a customer applies to all attendants for that customer.

Digital Private Signaling System 1 (DPNSS1) Executive Intrusion

If attendant secrecy is not active when the attendant attempts Executive Intrusion, the source is automatically excluded. If Enhanced Secrecy is equipped, source exclusion includes the removal of the Enhanced Secrecy warning tone when Executive Intrusion is activated.

Music

During secrecy, if there is only one undesired party in the conference, music is not provided to this party when excluded. However, intrusion tone is given to this party.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 96: LD 15 - Allow/deny Attendant Secrecy for a customer.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- OPT	(SYD) SYA	(Deny) allow Attendant Secrecy.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 49: Attendant Splitting

Contents

This section contains information on the following topics:

Feature description

Attendant Splitting allows the attendant to talk privately to the source or destination side of an existing connection on the console. The Exclude Source (EXCL SRC) key allows the attendant to speak privately with the destination (called) party. The Exclude Destination (EXCL DEST) key allows the attendant to speak privately with the source (calling) party.

Operating parameters

This feature is active only while the attendant is involved in the call.

Attendant Splitting applies to attendant consoles only.

Feature interactions

Attendant Recall

After the attendant and the two parties have been connected, the attendant can use the Attendant Splitting feature to communicate separately with either party.

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

To speak privately to the source party:

1. Press **EXCL DEST**.
2. To connect yourself, the caller, and the called party, press the **lpk** key.
3. To end your connection in the call, press **Rls**.

To speak privately to the destination party:

1. Press **EXCL SCR**.
2. To connect yourself, the caller, and the called party, press the **lpk** key.
3. To end your connection in the call, press **Rls**.

Chapter 50: Attendant Supervisory Console

Contents

This section contains information on the following topics:

Feature description

The Supervisory Console feature allows one attendant console in a customer group to function in a supervisory capacity when put into the Position Busy state. The elements of the Supervisory Console feature allow any of the following functions.

Attendant Status Display

The supervisor, by monitoring the attendant status display, can determine how many attendant positions are in service and able to receive calls.

If 1 to 20 attendants are assigned within a customer group, the supervisory console can monitor their status using Trunk Group Busy keys. No add-on module is necessary.

When an indicator on the module associated with a particular attendant is on, the attendant is available to service calls. If the indicator is off, the attendant position is in a Position Busy state. Attendant status indicators are only operable when the supervisory console is in a supervisory mode (Position Busy key operated). When the supervisory attendant is in Position Busy, the LED associated with the supervisor fast flashes at 120 ipm.

Attendant Status using Lamp Field Array

A supervisory console can have up to 49 status indicators when used in the Standard Busy Lamp Field mode. When using Enhanced Busy Lamp Field mode, a supervisory console can display the status of all attendant consoles in the customer group. Figure 4: Enhanced Busy Lamp Field Supervisory mode on page 360 shows an example of Supervisory monitoring in Enhanced Busy Lamp Field mode on the Busy Lamp Field/Console Graphics Module.

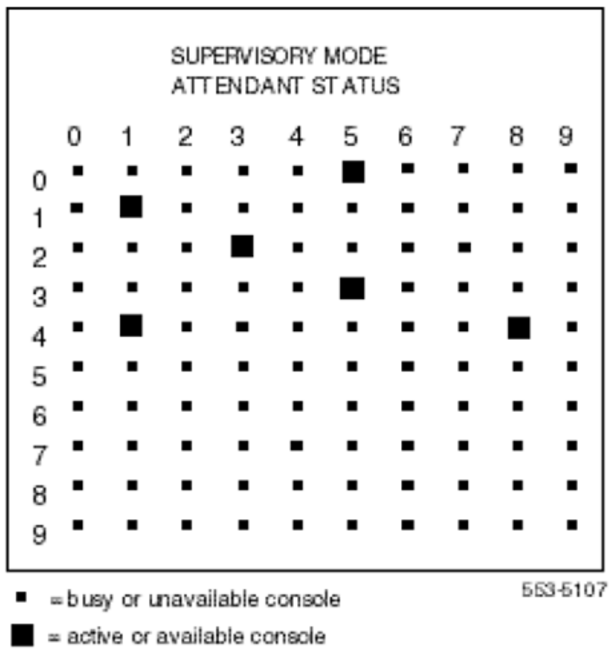


Figure 4: Enhanced Busy Lamp Field Supervisory mode

Visual indication of calls in queue

An attendant call queue holds incoming calls to the system that cannot be immediately answered by attendants. The supervisory console can monitor the call queue for specific types of incoming calls.

A maximum of 20 (ICI) key/lamp pairs can be assigned on an attendant console. Each ICI is assigned to handle a specific type of call (such as station, TIE, or dial 0) to the attendant. When a console is in the supervisory mode, the state of the lamp associated with each ICI provides a visual indication of the number of calls in the attendant queue for each ICI type. Each supervisory console ICI lamp state (dark, flash at 60 ipm, fast flash at 120 ipm, steadily ON) provides the supervisor with a visual indication of the number of calls in the queue for each call type. The ranges (calls in queue) are identified by one of three customer-specified thresholds that are set in service change programs.

Attendant Service Observation

This feature allows the supervisory attendant to monitor (listen only) calls in progress on other attendant loops without being heard. Service Observation requires the assignment of one key/lamp pair on the supervisory console flexible key strip. The key is assigned as Busy Verify through service-change programs. When the console is in Supervisory mode, the key function is Service Observation; when the console is operating as a normal attendant the key function is Busy Verify.

The observed attendant and the connected party or parties are not aware that their conversation is being monitored. The supervisor can release the connection by pressing the Release key. When the attendant is in a Service Observe mode, only the Release key is allowed as a valid input.

Supervisory assistance

An attendant can consult with, or transfer calls to, the supervisor or another attendant using the Interposition call feature. Interposition calls to the supervisor are allowed regardless of the mode of operation (Supervisory or Attendant). The supervisor can use the Interposition call feature to contact any attendant, except those in Position Busy. When the supervisor is conferring with an attendant, subsequent calls to the supervisor receive a busy indication.

If an attendant calls the supervisor who at the time is not in supervisory mode and is handling a call, the supervisory attendant interposition ICI lamp flashes at 60 ipm. As soon as the supervisor is idle, the calling attendant is connected to an idle loop on the supervisory console.

Interposition calls can be made from any attendant in the customer group to any other attendant within the customer group. Only one interposition call can be terminated on a console at a given time.

Supervisor serving as attendant

When the supervisor decides to act as an attendant, the supervisory console is removed from Position Busy. The system presents calls to the supervisory console as if it were a normal attendant console. The supervisory console must be idle to change states from attendant to supervisor or supervisor to attendant.

Operating parameters

The supervisory console and all attendant consoles (except Avaya 2250 Attendant Consoles) in the customer group must be assigned to QPC297 Attendant Console Monitor circuit packs. Their prime TN must be assigned to unit 0 and the secondary TN must be assigned to unit 1. Units 2 and 3 can be used for power; otherwise they must be left unassigned.

Important:

The Avaya 2250 Attendant Console must be a minimum vintage of AD and have the Attendant Supervisory Module (ASM) installed to allow supervision.

The supervisory console must have a Digit Display (DDS).

An Avaya 2250 Attendant Console equipped with a Busy Lamp Field/Console Graphics Module (BLF/CGM) can display the status of all attendant consoles (up to the maximum 63) by using the Enhanced Busy Lamp Field mode. The BLF/CGM must be minimum vintage AD to provide this capability.

One supervisory console can be assigned per customer. Only one attendant console (1 to 63) can be assigned as a supervisory console.

The customer group must be equipped with more than one attendant.

When using the Attendant Supervisory Module (ASM), the console TN must be configured on unit 0, 4, 8, 16, and so on. The secondary TN (SETN) unit must succeed the Primary TN (1, 5, 9, 17, and so on). The ASM TN is then configured with TYPE = PWR. The PWR TN must succeed the SETN (2, 6, 10, 18, and so on).

Feature interactions

Add-on modules

Add-on modules (key/lamp strips and lamp field arrays used to display attendant status) can be used for other purposes defined by the customer when the console is in Normal mode; however if the Busy Lamp Field is assigned to display attendant status, it cannot be used for other functions during any mode of the attendant console.

Attendant Administration

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

Attendant Position Busy

Activation of the Position Busy key on a Supervisory console puts the console in the supervisory mode.

Controlled Class of Service, Enhanced

When the attendant is in the supervisory mode, Controlled Class of Service programming is prohibited.

Console Presentation Group Level Services

The supervisory console specified for a customer belongs to one Console Presentation Group (CPG). In the Supervisory mode, ICI indicators show only the information for ICIs in that CPG. Thresholds specified in the Customer Data Block apply only to the CPG where that console resides, and do not effect any other CPG.

Departmental Listed Directory Number

The supervisory capabilities extend to all attendant consoles defined within the customer group. The attendant console serving as supervisor should be a member of every Departmental Listed Directory Number group so that it can serve all groups when operating in the Normal mode.

End-to-End Signaling

The supervisor can operate Attendant End-to-End Signaling (AEES) if there is a call on the active loop key. An attendant in AEES mode can be monitored by the supervisor.

Multi-Tenant Service

The supervisory capabilities extend to all attendant consoles defined within the customer group, regardless of tenant partitioning. The attendant console serving as supervisor should be a member of every Call Presentation Group so that it can serve all Tenant groups when operating in the Normal mode.

Source Included when Attendant Dials

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

Feature packaging

Supervisory Console (SUPV) package 93 has no feature package dependencies.

Feature implementation

Task summary list

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

1. Table 97: LD 15 - Enable/disable feature for an Avaya 2250 Attendant Console with a Console Graphics Module in the Standard Busy Lamp Field mode. on page 365
2. Table 98: LD 15 - Enable/disable feature for an Avaya 2250 Attendant Console with a Console Graphics Module in the Enhanced Busy Lamp Field mode. on page 365
3. Table 99: LD 12 - Enable/disable supervisory console Silent Observe. on page 366
4. Table 100: LD 12 - Enable/disable supervisory console for Avaya 2250 Attendant Consoles with Enhanced Busy Lamp Field and Silent Observe. on page 366
5. Table 101: LD 15 - Enable/disable an Avaya 2250 Attendant Console using Trunk Group Busy keys as status keys. on page 366

Table 97: LD 15 - Enable/disable feature for an Avaya 2250 Attendant Console with a Console Graphics Module in the Standard Busy Lamp Field mode.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- OPT	(XTG) ITG	Exclude/include Trunk Group Busy Indication.
- SPVC	1-63 0	Attendant number for supervisory console. No supervisory console.
- - SBLF	(NO) YES	Supervisory lamp field array is not or is to be used to monitor other attendant consoles.
- ITH1	1-255	Visual indication threshold 1 (number of calls in queue Š ITH1 but < ITH2).
- ITH2	2-255	Visual indication threshold 2 (number of calls in queue Š ITH2 but < ITH3).
- ITH3	3-255	Visual indication threshold 3 (number of calls in queue Š ITH3).

Table 98: LD 15 - Enable/disable feature for an Avaya 2250 Attendant Console with a Console Graphics Module in the Enhanced Busy Lamp Field mode.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and Avaya CS 1000E system.
- OPT	(XBL) IBL	Exclude/include Busy Lamp Field or Console Graphics Module.
- - SPVC	1-63 0	Attendant number for supervisory console. No supervisory console.
- ITH1	1-255	Visual indication threshold 1 (number of calls in queue Š ITH1 but < ITH2).
- ITH2	2-255	Visual indication threshold 2 (number of calls in queue Š ITH2 but < ITH3).
- ITH3	3-255	Visual indication threshold 3 (number of calls in queue Š ITH3).

Table 99: LD 12 - Enable/disable supervisory console Silent Observe.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	0 BVR	Add Busy Verify key (key 0) for silent observation.

Table 100: LD 12 - Enable/disable supervisory console for Avaya 2250 Attendant Consoles with Enhanced Busy Lamp Field and Silent Observe.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
EBLF	(BLFD) BLFA	(Deny) allow Enhanced Busy Lamp Field.
KEY	0 BVR	Add Busy Verify key (key 0) for silent observation.

Table 101: LD 15 - Enable/disable an Avaya 2250 Attendant Console using Trunk Group Busy keys as status keys.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- OPT	(XTG) ITG	Exclude/include Trunk Group Busy Indication.
- SPVC	1-63 0	Attendant number for supervisory console. No supervisory console.
- - SBLF	NO	Supervisory lamp field array is not to be used to monitor other attendant consoles.
- ITH1	1-255	Visual indication threshold 1 (number of calls in queue Š ITH1 but < ITH2).
- ITH2	2-255	Visual indication threshold 2 (number of calls in queue Š ITH2 but < ITH3).

Prompt	Response	Description
- ITH3	3-255	Visual indication threshold 3 (number of calls in queue Š ITH3).

Feature operation

Enable/disable Supervisory mode

To put your console in Supervisory mode, follow these steps:

1. Press the enable/disable supervisory modeicon when your console is idle (all lpk indicators are off). Yourconsole is now in Position Busy mode, preventing calls from ringingat your console.

2. To cancel Supervisory mode, press the enable/disablesupervisory mode icon again.

Monitor other attendants

In Supervisory mode, you can monitor selected attendant calls without being detected by either the attendant or the caller. To monitor an attendant, follow these steps:

1. Once in Position Busy mode, select an idle loop key.
2. Press **obs/B. ver.**
3. Dial the access code, then the attendant number:
   a. If the called attendant is talking to a caller, you hear the conversation but you cannot be heard.
   b. If the called console is idle, the S and D indicators go on.
   c. If the called console is in Position Busy mode, you hear a fast busy tone, the S and D indicators flash quickly, and the OBS/B. VER indicator goes off.
4. Press **Rls** to end the procedure.

Call an attendant

To call an attendant in your group, follow these steps:

1. Once in Position Busy mode, select an idle **Ipk** key.
2. Dial the attendant access code.
3. Dial the attendant code. You hear ringing. The S indicator flashes slowly.
4. Press **Rls** to end the call. The S indicator goes on steadily, and the RLS indicator goes on.

Transfer a call to an attendant

You can transfer a call to an attendant in your group, even if the attendant's console is in Position Busy mode. To transfer a call, follow these steps:

1. Dial the attendant access code; then the attendant code. The EXCL SRC indicator goes on; the caller is automatically placed on hold. The D indicator flashes slowly, the Ipk and S indicators are on.
    a. If you dial an incorrect attendant code or if the called console is in Night Service mode, the transfer cannot be completed. You hear a fast busy tone and the D indicator remains off. Press **Rls**.
    b. If the called console is busy, you hear a busy tone and the D indicator continues to flash slowly. Press **Rls** and your call is placed in the attendant queue.
2. Press the **Ipk** key when the attendant answers. The EXCL SRC indicator goes off and the D indicator lights steadily. You, the caller, and the attendant are connected.
3. Press **Rls** to end your connection in the call.

Assist an attendant

Even when your console is in Supervisory mode, an attendant can call you for assistance or transfer a call to you by following these steps:

1. You receive a call from an attendant while you are in Supervisory mode. You hear a tone. The S indicator flashes and the INTER POS. C. indicator goes on.
2. Press the **Ipk** key next to the flashing S indicator. The tone stops; the Ipk and S indicators light steadily. You are connected to the call.

Important:

If it is a transferred call, the Call Waiting indicator lights. You must exit Position Busy mode to answer the call.

Chapter 51: Attendant Trunk Group Busy Indication

Contents

This section contains information on the following topics:

Feature description

The attendant can control user access to a trunk route by pressing the appropriate Trunk Group Busy key. Station users with a Trunk Group Access Restriction (TGAR) from 0 to 7 accessing the route that has been busied out will be automatically intercepted to the attendant. Station users with a TGAR of 8 to 31 will not be affected and can dial out in the normal manner.

The Avaya 2250 Attendant Console can have up to 20 Trunk Group Busy keys.

Trunk Group Busy Indication is allowed or denied on a customer basis. If allowed, the lamps associated with the Trunk Group Busy keys will provide visual indication of the status of the trunks within the route (See Table 102: Lamp states of Trunk Group Busy keys on page 371).

Table 102: Lamp states of Trunk Group Busy keys

Lamp state	Status of trunks
Off	Some of the trunks in the route are idle.
Flashing	All of the trunks in the route are busy.

Lamp state	Status of trunks
Steadily lit	The attendant has taken control of the route.

Trunk Routes 0 to 9 are automatically assigned to keys 0 to 9 on the console.

On the Avaya 2250 Attendant Console, Trunk Routes are assigned to keys 0 to 9 and 10 to 19 when the Shift key is activated.

Operating parameters

There are no operating parameters associated with this feature

Feature interactions

Music

A music route that appears on a Trunk Group Busy key on the attendant console cannot be controlled by activation of the Trunk Group Busy key. In addition, the associated lamp will not reflect the status of the music trunks.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 103: LD 15 - Allow Trunk Group Busy keys.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options.

Prompt	Response	Description
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- OPT	(IC1) IC2	Allow Trunk Group Busy keys, where: IC1 = 10. IC2 = 20 for M2250.
- OPT	(XTG) ITG	(Exclude) include Trunk Group Busy Indicator keys.

Feature operation

To restrict access to a trunk route (make it busy to users):

- Press the **Trunk Group Busy** key associated with the trunk. The indicator goes on and remains steady.

To allow access to the trunk route:

- Press the **Trunk Group Busy** key associated with the trunk. The indicator goes off.

Chapter 52: Audible Reminder of Held Calls

Contents

This section contains information on the following topics:

Feature description

Occasionally, a user may forget that a call has been placed on hold. Audible Reminder of Held Calls (ARHC) allows an audible tone to operate as a reminder of a held call. It provides for a ring on analog (500/2500 type) telephones and a tone on proprietary telephones. The cadence and the duration between cadences are programmed per customer. This ability allows the user to differentiate between the cadence for Audible Reminder of Held Calls (ARHC) and the cadences of other existing features.

The station user will hear a ring or tone, which is repeated every 2 to 120 seconds depending on how this feature is programmed, as a reminder that a call is being held. A single-line telephone user must hang up after putting a call on Permanent Hold in order to start the timer.

Operating parameters

For analog (500/2500 type) telephones, Audible Reminder of Held Calls (ARHC) applies only to permanent hold. When using ARHC on a Meridian 1 proprietary telephone, the station user must not be originating, receiving, or active on another call.

Audible Reminder of Held Calls is supported on Multiple Appearance DNs; however, only the appearance initiating Hold will receive the reminder ring.

This feature does not operate on attendant consoles.

Feature interactions

Automatic Line Selection

The Audible Message Waiting signal is given if there is a message waiting on whatever line is selected by Outgoing Line Selection.

Call Hold, Permanent

Permanent Hold must be enabled in LD 10 for the single-line telephone; however, the ARHC timer takes precedence over the Permanent Hold timer.

On Hold on Loudspeaker

This feature works with the On Hold on Loudspeaker (OHOL) feature as for normal calls on hold (that is, gives a reminder there are calls on hold). Therefore, it is not recommended to use this feature with the OHOL feature.

Tones and Cadences

This feature allows for a definable cadence as a reminder of a held call. With an analog (500/2500 type) telephone, the cadence is determined by the customer's Flexible Tones and Cadence (FTC) table for the holding party. Ringing on an analog (500/2500 type) telephone is not affected by definitions for the Incoming Route option. The cadence for the reminder, and the duration between reminder rings, is always defined within the customer's tone table.

Feature packaging

This feature is included in base system software.

Feature implementation

Task summary list

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

1. Table 104: LD 15 - Set duration between reminder cadences for Audible Reminder of Held Calls. on page 377
2. Table 105: LD 10 - Allow/deny Audible Reminder of Held Calls for analog (500/2500 type) telephones. on page 378
3. Table 106: LD 11 - Allow/deny Audible Reminder of Held Calls for proprietary telephones. on page 378

Table 104: LD 15 - Set duration between reminder cadences for Audible Reminder of Held Calls.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	TIM	Timers.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- DBRC	2-(60)-120	Duration between reminder cadences for Audible Reminder of Held Call. An odd numbered entry is rounded up to the next even number.

Table 105: LD 10 - Allow/deny Audible Reminder of Held Calls for analog (500/2500 type) telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500 2500	500/2500 telephone type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(XFD) XFA (ARHD) ARHA	(Deny) allow call transfer. (Deny) allow Audible Reminder of Held Calls.
FTR	PHD	Permanent Hold allowed.

Table 106: LD 11 - Allow/deny Audible Reminder of Held Calls for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(ARHD) ARHA	(Deny) allow Audible Reminder of Held Calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 53: Authorization Code Security Enhancement

Contents

This section contains information on the following topics:

Feature description

The Authorization Code Security Enhancement feature enables a user to temporarily override the access restrictions assigned to a station or trunk because of their assigned Network Class of Service (NCOS), Class of Service (COS), and Trunk Group Access Restrictions (TGAR) codes. If a user requires access to system facilities in addition to that allowed on the telephone, the Authcode feature can be used to provide them.

The Authorization Code (Authcode) Security Enhancement feature alerts the technician when an invalid Authcode is entered by generating an Authcode Alarm. The Alarm indicates to the technician that a valid user has inadvertently dialed the wrong digits or some unauthorized person may be trying to use an Authcode to illegally access the switch.

The Authcode Alarm is generated upon detection of violation of all Authcode-related features (that is, Basic, Network, and Station Specific Authorization code), except for calls originated by the attendant.

An additional class of alarm (Security Administration—SECA) lets users distinguish security violations from other types of system messages. Once the system detects an invalid authorization code, it waits until the interdigit timout passes its limit, at which point the overflow tone is provided and the SECA001 alarm message displays on the TTY. However, if the user releases the call before the interdigit timeout reaches its limit, then the SECA001 alarm message does not display on the TTY. The SECA001 alarm message uses the following format:

- originated station or trunk Terminal Number
- Calling Line Identification (CLID) when the call is originated from an Integrated Services Digital Network (ISDN) trunk
- the Authorization code entered

Operating parameters

This feature is enabled through the Authcode data block in LD 88.

The Authcode Alarm feature does not apply to calls originated by an attendant.

All existing operating parameters relating to Authorization Code usage apply to this feature.

All existing operating parameters relating to Fault Management apply to this feature.

For security reasons, the SECA0001 alarm should not be configured in the Exception Filter table.

Feature interactions

Authorization Code Features

A Security Administration (SECA) message will be printed to the configured Maintenance Terminal (MTC), Filtered Alarm Output (FIL) console and/or the configured History File when an invalid Authcode is detected. The following features relate to Authorization Codes and are thus impacted: Basic Authorization Codes; Network Authorization Codes; Authcode Conditionally Last; Direct Inward System Access with Authorization Code; Station Specific Authcode; Speed Call/Autodial with Authorization Codes; Call Forward with Authorization Codes; Scheduled Access Restrictions with Authorization Codes; Network Queuing/Remote Virtual Queuing with Authorization Codes; Coordinated Dialing Plan with Authorization Codes; and Flexible Feature Code with Authorization Codes.

Charge Account, Forced

If the Authorization Code is used to change the Class of Service of the user, the new Class of Service must be TLD, CTD, or CUN. If an Authorization Code entered after FCA has altered the Class of Service to unrestricted (UNR), the change made by the Authorization Code still comes into effect.

If the originator's Network Class of Service (NCOS) has been changed by an Authorization Code prior to an applicable FCA entry, the new NCOS is replaced by the FCA NCOS, provided the new Facility Restriction Level (FRL) is not lower than the existing FRL. Similarly, if the originator's NCOS has been changed by an FCA entry, the NCOS will be changed again by a valid Authorization Code entry.

China - Flexible Feature Codes - Outgoing Call Barring

Digits dialed after an Authorization Code are checked against the active Outgoing Call Barring level.

Direct Private Network Access with Authorization Code Retry

Only when an Authcode retry fails will a SECA message be printed to the configured MTC, FIL console and/or the configured History File.

Last Number Redial

These codes are not stored in Last Number Redial (LNR). To use these features when calling the number stored in LNR, the code must first be dialed manually. When dial tone is returned, LNR can be used to complete the dialing.

New Flexible Code Restriction

If the Class of Service of the authorization code is Toll Denied (TLD), NFCR is applied. If the Class of Service is Conditionally Unrestricted (CUN) or Conditionally Toll Denied (CTD) and the call is not routed through BARS/NARS, CDP or ANI, NFCR is applied.

Pretranslation

The first digit dialed after a valid Authorization Code is sent to the pretranslator.

Scheduled Access Restrictions

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

Speed Call, System

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

Station Specific Authorization Code

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

Stored Number Redial

The Authorization code is not stored. To store a code, dial the code prior to using Stored Number Redial to dial the call.

Feature packaging

This feature is included in base system software.

The following software packages are optional, but may be needed depending upon the application:

- Alarm Filtering (ALRM_FILTER) package 243
- Basic Authorization Code (BAUT) package 25
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Coordinated Dialing Plan (CDP) package 59
- Direct Private Network Access (DPNA) package 250
- Direct Inward System Access (DISA) package 22

- Network Class of Service (NCOS) package 32
- Network Authorization Code (NAUT) package 63
- Station Specific Authcodes (SSAU) package 229
- Recorded Announcement (RAN) package 7
- Scheduled Access Restrictions (SAR) package 162
- System Speed Call (SSC) package 34, or Network Speed Call (NSC) package 39

Feature implementation

Task summary list

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

1. Table 107: LD 88 - Configure the Authcode Alarm for each customer. on page 383
2. Table 108: LD 117 - Configure the Alarm Filter. on page 384

Table 107: LD 88 - Configure the Authcode Alarm for each customer.

Prompt	Response	Description
REQ	NEW CHG	Configure or change.
TYPE	AUB	Authcode Data Block.
CUST	xx	Customer number, as defined in LD 15
SPWD	xxxx	Secure data password.
ALEN	1-14	Number of digits in Authcode.
ACDR	(NO) YES	(Do not) activate CDR for authcodes.
AUTHCOD_ALRM	(OFF) ON	(Disable) enable Authcode Alarm.
RANR		RAN route number for "Authcode Last" prompt (NAUT)
	0-511	Range for Large System and Avaya Communication Server 1000E system.

Table 108: LD 117 - Configure the Alarm Filter.

Command	Description
NEW EPT aa.. a INFO x Assign Information severity to new EPT entry, where: • aa... a = an event class with an event number (e.g. BUG1000, ERR0025, SECA0001) • x = optional entry to escalate value of EPT entry from (0)-Suppress value, as defined by default or your CHG SUPPRESS entry. or NEW EPT aa... a EDT x Assign NT-defined severity from EDT to new EPT entry, where: • aa... a = an event class with an event number (e.g. BUG1000, ERR0025, SECA0001) • x = optional entry to escalate value of EPT entry from (0)-Suppress value, as defined by default or your CHG SUPPRESS entry. or NEW EPT aa... a MAJOR x Assign Major severity to new EPT entry, where: • aa... a = an event class with an event number (e.g. BUG1000, ERR0025, SECA0001) • x = optional entry to escalate value of EPT entry from (0)-Suppress value, as defined by default or your CHG SUPPRESS entry. or NEW EPT aa... a MINOR x Assign Minor severity to new EPT entry, where: • aa... a = an event class with an event number (e.g. BUG1000, ERR0025, SECA0001) • x = optional entry to escalate value of EPT entry from (0)-Suppress value, as defined by default or your CHG SUPPRESS entry. or NEW EPT aa... a CRITICAL x Assign Critical severity to new EPT entry, where: • aa... a = an event class with an event number (e.g. BUG1000, ERR0025, SECA0001) • x = optional entry to escalate value of EPT entry from (0)-Suppress value, as defined by default or your CHG SUPPRESS entry.	

Feature operation

No specific operating procedures are required to use this feature.

Chapter 54: Autodial

Contents

This section contains information on the following topics:

Feature description

Autodial (ADL) allows users to dial a number by pressing a single key. Proprietary telephones and attendant consoles can be assigned an Autodial key/lamp pair.

The number stored against the Autodial key can be programmed or changed at any time. The maximum number of digits the user is allowed to program can be 4, 8, 12, 16, 20, or 23 digits. Depending on the length allowed, the Autodial number can be another DN or an access code plus further digits. The asterisk (*) can be used as a pause for outpulsing (that is, for outgoing trunks) when required. When the Autodial key is pressed, the stored number is processed as if it had been dialed manually.

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/Autodial with Authorization Code. This enhancement allows an Authorization Code to be included in a Speed Call entry or an Autodial key. Entries can contain any one of the following combinations:

- SPRE code + digit 6 + authorization code
- SPRE code + digit 6 + authorization code + #, or
- SPRE code + digit 6 + authorization code + # + Electronic Switched Network (ESN) access code and dialed number.

Autodial Flexible Feature Codes A user can define an Autodial DN that is automatically dialed by the system in one of two ways:

- In LD 10, while defining the Autodial DN length under the feature (FTR) ADL.
- Using the Autodial Activate (ATDA) FFC, defined in LD 57. This method requires that the length of the Autodial must first be defined in LD 10. The user goes off hook and dials the ATDA FFC. Upon receiving dial tone, the user enters the desired Autodial DN, and then goes on hook.

If, after going off hook, no digits are entered within a customer-defined period of time (defined in LD 15) under ADLD (Autodial Delay), the Autodial DN is automatically dialed.

In LD 10, the user can define a partial DN as an Autodial DN. The user can enter the remaining digits while making a call – the user goes off hook, waits for the dial tone to time out, and then enters the remaining digits of the desired DN. The call is then dialed out.

To deactivate Autodial, the user dials either the Autodial Deactivate (ATDD) FFC (defined in LD 57) or the general Deactivate (DEAF) FFC (also defined in LD 57).

Operating parameters

Autodial must be assigned to a key/lamp pair. As a result, it is not available on analog (500/2500 type) telephones.

To use Autodial, the Autodial Activate (ATDA) FFC must have been entered previously and an Autodial number must be stored.

An attendant can enter an Authorization Code for other callers provided that the system is equipped with the Network Authorization Code (NAUT) package.

On attendant consoles, pressing the Autodial key, then pressing a Speed Call key is not allowed.

Authorization Code Conditionally Last is not supported by the Autodial feature.

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

The Autodial feature allows a maximum of 23 digits including the SPRE code, the digit 6, the Authorization Code, the delimiter (#), the ESN access code, and the dialed number.

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

A Meridian 1 proprietary digit display telephone can display up to 16 digits. Additional digits cause the digits to scroll off the display.

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

The Authorization Code is not validated during the storing process. An invalid Authorization Code is detected when the Autodial key is activated.

Network Automatic Route Selection (NARS) and Basic Alternate Route Selection (BARS) does not support the asterisk (*) as a pause when dialing an autodial number.

Feature interactions

AC15 Recall: Transfer from Meridian 1

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

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

Automatic Redial

Automatic Redial can be activated on a dialed number using the Autodial (ADL) key.

Call Forward and Busy Status

Party A can use the Busy/Forward Status (BFS) key as an Autodial key to dial party B.

Call Party Name Display

No name information displays during the programming of Autodial numbers.

Calling Party Privacy

An outgoing trunk call initiated by pressing the Autodial key will carry the Privacy Indicator if the Calling Party Privacy (CPP) code followed by the normal dialing sequence is stored against the Autodial key. The CPP code is counted against the maximum number of digits (currently 23) stored against the Autodial key.

A user can also store the CPP code against the Autodial key. An outgoing CPP call can be initiated by pressing the Autodial key, followed by manually dialing the digits.

An outgoing CPP call can also be initiated by dialing the CPP code, followed by pressing the Autodial key against which the normal dialing sequence of digits have been stored.

Charge Account and Calling Party Number

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

Charge Account, Forced

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

China - Flexible Feature Codes - Busy Number Redial Enhanced Flexible Feature Codes - Busy Number Redial

Activation of Busy Number Redial (BNR) changes the activation of Autodial. The DN that is auto dialed becomes the DN that was busy. When the BNR activation timer expires or the busy DN is redialed when it is idle, the autodial capability is deactivated, but the number saved is not cleared. If Autodial is then activated without entering a DN, the number used is the formerly busy DN.

Activation of Autodial when BNR is active deactivates BNR.

China Number 1 Signaling Enhancements

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

Dial Intercom

The Dial Intercom code can be dialed using Autodial or Speed Call.

Direct Private Network Access

If Autodial is programmed with a valid Authcode for Authcode Last component of Direct Private Network Access followed by an octothorpe "#", the existing Authcode Last operation will reject the Authcode as an invalid Authcode. If Authcode Last Retry is defined, the caller will be prompted for the Authcode again.

Flexible Hot Line Enhanced Hot Line

Flexible Hot Line and/or Enhanced Hot Line are mutually exclusive with the Autodial feature.

Intercept Computer Dial from Directory

It is possible to press the Autodial (ADL) key (in which some digits are stored such as an Electronic Switched Network (ESN) code or Flexible Feature Code (FCC)), and then dial a DN from the Intercept Computer. The DN will then be stored on the ADL key.

Last Number Redial

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

Station Specific Authorization Code

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

Speed Call Delimiter

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

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

Autodial on a E3W trunk will fail for toll calls. The reason is that E3W trunks do not wait for the ANI request from the Public Exchange/Central Office, which is expected to appear after the toll access code is dialed. The Public Exchange then does not accept the call due to failure to receive ANI information.

User Selectable Call Redirection

User Selectable Redirection Allowed (USCR) does not support Autodial; Autodial cannot be used to dial all or part of the digits for USCR programming.

Feature packaging

Optional Features (OPTF) package 1 includes Autodial and has no feature package dependencies.

To implement Autodial with Authorization Code, the following packages are required:

- Charge Account/Authorization Code Base (CAB) package 24, or Basic Authorization Code (BAUT) package 25, or Network Authorization Code (NAUT) package 63.
- Optional Features (OPTF) package 1, or System Speed Call (SSC) package 34, or Network Speed Call (NSC) package 39.

The following packages are required for Autodial FFCs:

- Flexible Feature Codes (FFC) package number 139, and
- Background Terminal Facility (BGD) package 99.

Feature implementation

Task summary list

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

1. Table 109: LD 11 - Assign Autodial key for proprietary telephones. on page 391
2. Table 110: LD 12 - Assign Autodial key for Avaya 2250 Attendant Console. on page 391
3. Autodial Tandem Transfer on page 395

Table 109: LD 11 - Assign Autodial key for proprietary telephones.

Prompt	Response	Description
REQ:	CHG	Change.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx ADL yy zzz...z	xx = assigned key number. yy = the length of the Autodial number (4, 8, 12, 16, 20, or 23 digits; default is 16). zzz...z = the digits to be dialed automatically (optional).

Table 110: LD 12 - Assign Autodial key for Avaya 2250 Attendant Console.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	2250	Attendant console type.
TN		Terminal number
	l s c u	Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Prompt	Response	Description
KEY	xx ADL zzz...z	xx = assigned key number. zzz...z = the digits to be dialed automatically (optional).

Table 111: LD 15 - Define Autodial Delay in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add or change.
TYPE:	FFC	Flexible Feature Codes
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		
- ADLD	(0)-20	Autodial Delay, in seconds. If 0, then FFC Autodial for 500/2500 telephones is disabled. Only prompted if FFC package (139) is equipped. Inputs are rounded up to the next valid increment of two (that is, input of 11 would be rounded up to 12).

Feature operation

To program Autodial, follow these steps:

1. While the handset is on hook, press the **Autodial** key. The associated lamp flashes.
2. Dial the desired number and press the **Autodial** key again. The lamp goes dark.

To use Autodial, follow these steps:

1. Lift the handset off hook, or press the **Handsfree** key if allowed.
2. Press the **Autodial** key. The call is dialed.

The following instructions are for using the Autodial FFCs:

- Activate and program The user must dial the Autodial Activate (ATDA) FFC followed by the number to be stored as the Autodial number.
- Activate only The user must dial the Autodial Activate (ATDA) FFC.
- Deactivate The user must dial the Autodial Deactivate (ATDD) FFC or the Deactivate (DEAF) FFC.
- Use The user goes off hook, if no digits are dialed within the customer defined time period (ADLD), the system then dials the number stored as the Autodial number.

Note:

To use Autodial, the Autodial Activate (ATDA) FFC must have been entered previously and an Autodial number must be stored.

Chapter 55: Autodial Tandem Transfer

Contents

This section contains information on the following topics:

Feature description

Prior to the introduction of this feature, in order to access the Central Office (CO) transfer feature after a Centrex/Trunk Hook Flash on an established trunk call, the user had to manually dial the digits. This procedure permits call completion, but is slow and requires knowledge of the full telephone number. The Autodial Tandem Transfer (ATX) feature allows the Autodial key to be used after a switchhook flash to out pulse Dual-tone Multifrequency (DTMF) digits while a call is in an established state.

One application for the Autodial Tandem Transfer feature is for use in a 911 environment to transfer an emergency call from a Public Safety Answering Point (PSAP) to the most appropriate participating emergency agency. Manually dialing the digits by the PSAP in order to transfer the 911 call to another PSAP can take time and is subject to misdialing. To avoid this, the ADL key programmed with the special station number can be used to send digits to the tandem/Centrex office to transfer the call. Using the ATX feature, a PSAP can transfer the incoming call by pressing the Trunk Hook Flash (THF) key, waiting for a broken dial tone, and then pressing the ADL key.

Operating parameters

The Centrex/Trunk Switchhook feature only supports voice calls. Subsequently, the ATX feature which uses Centrex/Trunk Hook Flash does not support data calls.

Centrex/Trunk Hook Flash cannot be activated during Conference and No Hold Conference calls. Subsequently, the ATX feature which uses Centrex/Trunk Hook Flash does not support them either. Only two-party calls are supported by the ATX feature.

The following trunk types are supported by the ATX feature: AID, CAA, CAM, COT, TIE (supports ATX, not Trunk Hook Flash), CSA, DID, DOD, WATS, DTI, and DTI2.

The ATX feature is not supported on analog (500/2500 type) telephones, attendant consoles, and BRI telephones.

End-to-End signaling (EES) is not supported for this feature (only Improved End-to-End signaling is supported).

Single CPU machines are not recommended for 911 applications. Meridian 911 hardware may be required for 911 applications.

Feature interactions

Automatic Dial

The ADL key is used by the Automatic Dial feature to send DN digits out during the dialing stage. Some of the digits, such as "#" and "*", have special meanings. The "*" causes a three-second pause, while the "#" means end of dialing.

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.

In the ATX feature when the ADL key is used during an established call, the DTMF tones corresponding to the digits programmed in the ADL key are sent out (using End-to-end Signaling to send the digit out). Therefore, the DTMF tones corresponding to "#" and "*" are outpulsed.

Call Detail Recording

No modifications to this feature are required for the ATX feature.

For 911 applications, most of the calls are incoming calls. The outgoing End-to-End Signaling digits are captured for incoming 911 calls on the incoming CDR records. This only applies to 911 trunks.

Centrex Switchhook Flash

Because Autodial Tandem Transfer uses Centrex Switchhook Flash (THF), it is affected by any modification to the THF enhancement feature.

Conference

The ATX feature is blocked during Conference and No Hold Conference calls.

Digit Display

Digit Display allows the automatic display of information relevant to normal call processing if the telephones have display capability and the Class of Service is ADD or DDS. When the THF key is pressed, the display gets cleared, and pressing the ADL key causes the ADL digits to be displayed. However, no ADL digits will be displayed if no Tone and Digit Switch (TDS)/XCT is available to generate the Dual-tone Multifrequency (DTMF) tones for the ADL digits.

End-to-End Signaling

EES is used to send the Automatic Dialing (ADL) digits to the Public Exchange/Central Office (CO). With Autodial Tandem Transfer (ATX), the 911 agent can use the ADL key or manually dial the digits, or use a combination of both methods, to dial the third party's number. The ADL key can be pre-programmed with a prefix and the remaining digits can be dialed manually to distinguish between different numbers. When you combine manual dialing with the ADL key, if EEST = YES and DTMF = YES in LD 15, you hear the DTMF feedback tone as a result of manual dialing and a single feedback tone as a result of pressing the ADL key. To get uniform feedback tone when using the ADL key along with manual dialing, set the DTMF prompt to NO in LD 15.

Improved End-to-End Signaling is used to send the pre-programmed ADL digits to the CO. With the ATX feature, a 911 Agent can use the ADL key, or manually dialed digits, or a combination of both to dial the third party's number. It is recommended to set the DTMF prompt to NO (EES – LD 15) to get uniform feed back tone (single feed back tone) when using the ADL key along with manual dialing.

Last Number Redial

Normally, when the ADL key is pressed during the dialing stage, the ADL number will replace the Last Number Redial number. In the ATX feature, however, when the ADL key is used during the established stage, the ADL digits will not substitute the Last Number Redial number.

Malicious Call Trace - Enhanced

Enhanced Malicious Call Trace implements the ability to send a call trace request to the CO and provides the possibility to record the call using a recorder. This feature also uses the Centrex/Trunk Switchhook Flash feature; the same enhancement applies to the ATX feature.

Speed Call

The Speed Call key cannot be used after THF or during an established call to send digits out to the far site; it can only be used during the dialing stage.

Feature packaging

Autodial Tandem Transfer (ATX) is package 258.

The following packages are also required:

- End-to-End Signaling (EES) package 10
- Trunk Hook Flash (THF) package 157

Feature implementation

Task summary list

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

1. Table 112: LD 11 - Define THF and ADL keys for Meridian 1 proprietary telephones on page 399
2. Table 113: LD 14 - Define THF Class of Service THFA for the trunk on page 399
3. Table 114: LD 15 - Define feedback tone when ADL digits are sent out on page 399

Table 112: LD 11 - Define THF and ADL keys for Meridian 1 proprietary telephones

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	...	
KEY	xx THF	Key xx is configured for the Centrex/Trunk Switchhook Flash feature.
	yy ADL ll zz..zz	Key yy is configured for the Autodial key; ll is the length of the autodial number (the default is 16). zz..zz are the digits to be dialed automatically.

Table 113: LD 14 - Define THF Class of Service THFA for the trunk

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	aaa	Trunk type, where: aaa = AID, CAA, CAM, COT, TIE (supports ATX, not Trunk Hook Flash), CSA, DID, DOD, WATS, DTI, and DTI2.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(THFD) THFA	The THF feature is (denied) allowed; the default is THFD.

Table 114: LD 15 - Define feedback tone when ADL digits are sent out

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	CDR	Call Detail Recording.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.

Prompt	Response	Description
CDR	YES	Call Detail Recording.
- ECDR	YES	Include EES digits in CDR record. This will include ADL digits that are outpulsed during an established call.
TYPE:	FTR	Features and Options.
EEST	YES NO	End-to-end Signaling feedback tone to originating party.
- DTMF	YES NO	Single feedback tone is provided for the user. With a Yes or No response, single tone feedback is only available.

Feature operation

Normal operation

1. An incoming call from a Central Office (CO) terminates to a proprietary telephone.
2. The telephone user presses the **THF** key, waits for a broken dial tone from the CO, and then presses the **ADL** key to send a string of digits to the CO (the ADL has been pre-programed with the number).
3. The CO will transfer the call to the third party dialed by telephone A.

Meridian 911 operation

1. An incoming 911 trunk call to a tandem/Centrex office terminates to a PSAP on the system.
2. The PSAP call taker presses the THF key, waits for a broken dial tone, and then presses the ADL key to call the proper number (such as a police station).
3. The PSAP call taker then disconnects to complete the transfer.

Chapter 56: Automatic Answerback

Contents

This section contains information on the following topics:

Feature description

Automatic Answerback (AAB), when assigned to a Meridian 1 proprietary telephone, allows any incoming call to a single appearance Prime Directory Number (PDN) to be answered automatically. An incoming call will ring one time, then the system will turn on Handsfree and establish a speech path. When either party hangs up, the call is automatically disconnected.

Automatic Answerback can be permanently assigned either as a Class of Service, or with an Automatic Answerback key/lamp pair assigned to allow activation/deactivation of the feature. If privacy is desired during a call, handset operation is allowed.

Operating parameters

This feature is available on the following telephones:

- 2002P1 and 2002P2
- 2004P1 and 2004P2
- 2007

- 2016
- 2033
- 2050PC
- 2210
- 2211
- 2212
- 2616
- 1120
- 1140
- 1150
- 1165
- 1210
- 1220
- 1230
- 3902
- 3903, 3903H, and 3903V
- 3904, 3904H, and 3904V
- 3905

Incoming ground start trunks must provide Answer Supervision. If not, the call is connected to the attendant who provides the necessary supervision.

The Prime DN (PDN) must be a single appearance DN.

Calls presented to DNs other than the PDN, or calls presented to the PDN when active on another DN, will not receive Automatic Answerback treatment.

Automatic Answerback can be provided as a Class of Service or on a key/lamp pair. You cannot assign both in service change.

Feature interactions

Automatic Line Selection

Automatic Answerback operates only on the Prime DN (key zero) and has no interrelation with Incoming Ringing/Non-Ringing Line Selection.

Called Party Disconnect Control

Incoming calls on a trunk with Called Party Disconnect Control Allowed that terminate on a telephone with Handsfree Answerback are answered automatically. They are not disconnected automatically, however, when the calling party goes on-hook.

Collect Call Blocking

The Automatic Answerback (AAB) feature, when assigned to a Meridian 1 proprietary telephone, allows any incoming call to a single-appearance Prime Directory Number (PDN) to be answered automatically. If an incoming DID or CO call terminates on a telephone with the AAB feature enabled, the call is automatically answered after one ring. If the telephone has a CCBA Class of Service, the CCB answer signal is provided in the place of the regular answer signal.

Hot Line

The Automatic Answerback feature is fully compatible with a two-way Hot Line key assigned as the Prime DN.

Message Center

If a telephone is in the Automatic Answerback mode, incoming calls are not routed to the Message Center.

Feature packaging

Automatic Answerback (AAB) package 47 has no feature package dependencies.

Feature implementation

Task summary list

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

1. Table 115: LD 11 - Assign Automatic Answerback as a Class of Service to supported telephones. on page 404
2. Table 116: LD 11 - Assign Automatic Answerback key to supported telephones. on page 404

Table 115: LD 11 - Assign Automatic Answerback as a Class of Service to supported telephones.

Prompt	Response	Description
REQ:	CHG	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 Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(HFD) HFA (AAD) AAA	(Deny) allow Automatic Answerback for all calls. AAA cannot be entered if the AAK key is already programmed. (Deny) Allow Handsfree.

Table 116: LD 11 - Assign Automatic Answerback key to supported telephones.

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

Prompt	Response	Description
	l s c u	Format for Large System and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(HFD) HFA	(Deny) allow Handsfree. HFA is allowed for the M2216 only.
	(AAD) AAA	Allow (Deny) Automatic Answerback. Must disable to add the AAK key.
KEY	xx AAK	Add Automatic Answerback key. xx = key number. The M2216 with AAA cannot use key 5 as a feature key. Key 5 is reserved for handsfree.

Feature operation

To activate Automatic Answerback, follow this step:

- Press **Auto Answer**. Incoming calls to your PDN will ring once, then be answered with Handsfree turned on.

To deactivate Automatic Answerback, follow this step:

- Press **Auto Answer**. Incoming calls to your PDN will not be answered automatically.

If Automatic Answerback is assigned as a Class of Service instead of a key on your telephone, you cannot deactivate it.

Chapter 57: Automatic Call Distribution

Automatic Call Distribution (ACD) is an optional feature. The ACD feature is used when a large number of incoming calls are answered by a group of ACD-assigned telephones. Incoming calls are served on a first-in, first-out basis and are distributed among the available telephones so that the agent position that has been idle the longest is provided with the first call. This guarantees that incoming calls are distributed equally to all agents.

For more information about the ACD feature, see *Avaya Automatic Call Distribution Fundamentals* (NN43001-551).

Chapter 58: Automatic Gain Control Inhibit

Contents

This section contains information on the following topics:

Feature description

The Automatic Gain Control (AGC) function, supported by the A44 chip in Meridian digital telephones, lowers handset sound levels to minimize background noise. The AGC Inhibit enhancement allows a customer to suppress this function, on a system basis.

Whenever a transmission download occurs, which happens following a SYSLOAD or when the telephone line cord is plugged in, the option setting in LD 17 is included in the message. The message is interpreted by telephone's firmware and the appropriate setting is applied.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 117: LD 17 - Define the AGC setting.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ATRN	Aries Transmission.
ATRN	YES	Aries (Meridian Modular telephone) transmission parameter; only prompted if the response to TYPE is CFN.
- AGCD	(NO) YES	Automatic Gain Control Disable.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 59: Automatic Guard Detection

Contents

This section contains information on the following topics:

Feature description

This feature verifies the transition from a high-resistance to a low-resistance loop upon correct seizure of an inactive trunk. Incorrect seizure results in the release of the faulty trunk and the attempted seizure of the next trunk in the hunt sequence.

Automatic Guard Detection will prevent the seizure of a trunk if the trunk:

- is an open circuit in tip, ring, or both; or
- has no current present when the trunk is seized

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

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

Feature implementation

Table 118: LD 14 -Enable or Disable Automatic Guard Detection for outgoing trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change data.
TYPE	a...a	Type of truck.
SEIZ	(NO) YES	Automatic Guard Detection for outgoing trunks (disabled) enabled.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 60: Automatic Hold

Contents

This section contains information on the following topics:

Feature description

The Automatic Hold feature allows an active call to be put on hold without having to use a separate Hold key. There are three ways to put a call on hold with the Automatic Hold feature:

- Press the active call key. The established call is automatically placed on hold.
- Press an idle Directory Number (DN) key. The established call is automatically placed on hold.
- Press any idle key and the established call is placed on hold.

If a set user is on an established call and wishes to answer an incoming call or initiate an outgoing call, the set user can press any idle DN key to place the call on hold and either initiate or establish a call on the same key. To terminate a call with the Automatic Hold feature, the Release key must be pressed.

This feature requires a new Class of Service implementation (Automatic Hold Class of Service).

Operating parameters

The Automatic Hold feature can be equipped on all multi-line proprietary sets. The functionality to hold a call already exists on the attendant console. The Automatic Hold feature is not applicable on analog (500/2500) type sets.

Feature interactions

Attendant Break-In to Inquiry Calls

A consultation call on proprietary sets, using a second DN along with Automatic Hold, is not treated as an inquiry call. The consultation call may be broken-in to, but the call held on the first DN is not involved in the Break-In.

Attendant Recall with Splitting

Automatic Hold does not have precedence over Attendant Recall (for instance, Automatic Hold cannot be activated until the attendant answers the recall presented on the console). However, it can be activated before the attendant answers a call transferred to the console.

Automatic Answer Back

The Automatic Hold feature is not applicable with the Automatic Answer Back feature.

Automatic Call Distribution Incalls Key

Automatic Call Distribution (ACD) does not override the Incall 5 key. The Incalls key is unique to the Automatic Hold feature. If an Automatic Call Distribution (ACD) agent has an active call on the Independent Directory Number (IDN) key, and a call comes in to an Incalls key, pressing the Incalls key to answer the call puts the active call on the IDN key on hold.

Call Transfer

If a call is established or ringing on the Transfer key, pressing any idle DN key automatically puts the call on hold. To transfer an active call, press the transfer key once to reestablish the call, press a second time to complete the transfer. To release the transfer feature you must press the release key.

Call Waiting

Pressing the Call Waiting key to answer a waiting call, makes that call active while the previous call is put on hold.

Called Party Control on Internal Calls

If Called Party Control on Internal Calls is active, an established call is put on hold regardless of the Automatic Hold Class of Service assigned to the set when a second call is originated or answered.

Conference

If a call is established on the conference key, pressing any DN key puts the Conference call on hold. The user must press the conference key to reestablish the call. Pressing the conference key a second time completes the Conference call.

Digit display

Digit display is the same with automatic hold as it was with manual hold.

Display Overflow on Calling Number Identification

If the number of Calling Number Identification (CNI) digits exceeds the capacity of the digit display, the active DN key can be pressed to show the remaining digits. If the active DN key is pressed again, the established call is placed on hold. The established call can be placed on hold, before the digits are displayed, by pressing any other DN key.

Enhanced Hotline and Hotline No Hold Conference

On proprietary sets pressing a designated Hotline key places an outgoing call to a pre-defined DN. Pressing any idle DN key or pressing the hotline key a second time can place this call on hold. The user can use the same DN key they used to put the call on hold to make an outgoing call or to answer an incoming call.

On a two-way Hotline key, the incoming call is held if the hotline key is pressed twice or if an idle DN key is pressed. Pressing the Release key while on an active Hotline call terminates the call.

The Conference-Hot Line (CH) key does not support Automatic Hold.

Group Call (GRC)

Only the originator of a Group Call (GRC) can put the Group Call on hold.

Hold Key

A set configuration with Automatic Hold Allowed Class of Service can still place calls on hold using the Hold key.

Individual Hold Enhancement

When a Multiple Appearance Directory Number (MADN) call is put on hold on proprietary sets, the Hold key lamp flashes at this user's set, while a slow flicker is shown at all other appearances of the same DN. With more than one single line MADN (SCR/SCN/HOT/PVR/PVN) active on a conference call, the user is put on hold either by pressing the Hold key, or with Automatic Hold feature enabled, the user can press the active single line MADN. With the Release option disabled, the active call on the single line MADN is put on hold. With the Release option enabled, the active call on the single line MADN is dropped.

Lamp Status

The LED lamp status indications of calls put on automatic hold are identical to those for calls that are put on hold using the Hold key.

Last Number Redial (LNR)

A set with Last Number Redial Allowed (LNA) Class of Service can put an active call on hold by pressing another idle DN key and still activate the Last Number Redial feature to make an outgoing call. Automatic Hold does not override this feature.

Music on Hold

Music on Hold can be applied to calls put on automatic hold.

No Hold Conference

The Automatic Hold feature does not apply in the case of a No Hold Conference call. Automatic Hold does not override the No Hold Conference feature.

Voice Call

If a user presses the Voice Call key while a call is established on the key, the call is placed on hold. If the Voice Call key is pressed while a call is established on another DN, the established call is put on hold.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 119: LD 11 - Allow or deny the Automatic Hold Class of Service for proprietary sets.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.

Prompt	Response	Description
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15.
CLS	AHA (AHD)	AHA = Automatic Hold allowed. (AHD) = Automatic Hold (denied).
KEY	xx aaa yyyy	The set type must be configured with two DN keys. Where: xx = key number aaa = DN key type. DN types supported include: ACD, CWT, DIG, GRC, HOT, MCN, MCR, SCR, SCN, or VCC. yyyy = Directory Number for key type. Refer to feature interactions in this chapter when assigning keys to see if feature operation conditions are affected.

Feature operation

Put a call on hold

With Automatic Hold enabled, a call can be placed on hold by pressing the DN on which the call is active or by pressing any other idle DN key.

Make a new call

An active call can automatically be placed on hold, if any idle DN key is pressed. A new call can now be made on the DN key that was pressed or any other DN key.

Answer a call

If the set user is on an active call and a second call is presented on another DN, the user can answer the incoming call which automatically places the first call on hold.

A user of a set having Automatic Hold Class of Service can still place an active call on hold by pressing the Hold key.

Terminate a call

To terminate a call the set user must press the Release key.

Chapter 61: Automatic Line Selection

Contents

This section contains information on the following topics:

Feature description

Automatic Line Selection allows manual or automatic selection of incoming and outgoing lines for a given Meridian 1 proprietary telephone on a Class of Service basis. When a user lifts the handset, the telephone automatically selects a preferred line according to its priority. The line preferences are as follows, listed in order of selection priority:

- Manual Line Selection

  The user manually selects the DN to be used before going off-hook. Dial tone is returned if the line is idle. If the line is ringing, the call is answered and connected to the speaker of the telephone or Handsfree unit.

- Incoming Ringing Line Selection

  With Incoming Ringing Line Selection enabled, when the user goes off-hook, the telephone automatically scans the DN keys (without the user first manually selecting a DN key). If a line on the telephone is ringing, it is selected and the call is answered.

- Incoming Non-Ringing Line Selection

With Incoming Non-Ringing Line Selection enabled, when the user goes off-hook, the telephone scans the DN lines and answers any unanswered incoming calls that appear but do not ring at that telephone.

- Outgoing Line Selection

  With Outgoing Line Selection enabled, when the user goes off-hook, the telephone scans the DN keys for an idle line. If a line is idle, it is selected and dial tone is returned.

- Prime Line Selection

  When the handset is lifted, the system processes any manual, incoming, or outgoing line selections. If no line is selected by one of these modes, a designated Prime Line (the DN on key 0) is selected.

Operating parameters

The Automatic Line Selection feature is available on Meridian 1 proprietary telephones only.

The user determines which line is in use by observing lamp state changes.

Feature interactions

Audible Message Waiting

The Audible Message Waiting signal is given if there is a message waiting on whatever line is selected by Outgoing Line Selection.

Automatic Call Distribution (ACD)

An ACD DN is not selected by automatic Incoming Non-Ringing and Outgoing Line Selection. It is selected by Incoming Ringing Line Selection.

Automatic Answerback

Automatic Answerback operates only on the Prime DN (key zero) and has no interrelation with Incoming Ringing/Non-Ringing Line Selection.

Automatic Redial

Manual Line Selection, Outgoing Line Selection or Prime Line Selection is interpreted as accepting the Automatic Redial (ARDL) by the calling party.

Call Waiting

A call on the Call Waiting key is not selected.

Dial Intercom

A Dial Intercom DN is selected by Incoming Ringing Line Selection and Outgoing Line Selection.

Group Call

This feature is not selected for automatic Outgoing Line Selection or Non-Ringing Line Selection. It is selected for Incoming Ringing Line Selection.

Hot Line

Since the Hot Line key acts as a Single Call Ring (SCR) key, incoming ringing line preference can be applied. Outgoing line preference automatically selects a line other than the current Hot Line, so that a Hot Line call is not accidentally activated

Private Line Service

A Private line DN is selected by Incoming Ringing/Non-Ringing Line Selection and Outgoing Line Selection.

Voice Call

This feature is not selected by automatic Outgoing Line Selection. It is selected for Incoming Ringing and Non-Ringing Line Selection.

Feature packaging

Automatic Line Selection (LSEL) package 72 has no feature package dependencies.

Feature implementation

Table 120: LD 11 - Assign Automatic Line Selection for each Meridian 1 proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	(IRD) IRA (NID) NIA (OLD) OLA	(Deny) allow incoming ringing line preference. (Deny) allow incoming non-ringing line preference. (Deny) allow outgoing line preference.
LPK	xx	Specify the last key to be scanned for line preference (such as 0-7, 10-17, 20-27). Prompted only if CLS = IRA, NIA, or OLA. **Note:** A value of 0 (zero) for LPK disables this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 62: Automatic Number Identification

Contents

This section contains information on the following topics:

Feature description

The Automatic Number Identification (ANI) feature automatically identifies a station originating an outgoing toll call and its destination party and transmits the information to a recording office.

A system with ANI sends information about stations involved in an outgoing toll call, using Multifrequency (MF) signaling, over Central Automatic Message Accounting (CAMA) trunks to toll-switching CAMA, Traffic Operator Position System (TOPS) or Traffic Service Position System (TSPS) offices.

The software portion of ANI performs the following functions:

- identifies an originating outgoing toll call
- determines the calling station identification, and controls the signaling and supervision of the ANI trunk circuit
- connects the MF sender and the ANI trunk circuit
- loads up to 16 digits that are to be MF outpulsed over the ANI trunk into the MF sender

- orders initiation of the outpulsing
- removes the connection between the trunk and the MF sender and establishes the speech path to the trunk

With the E.164/ESN Numbering Plan Expansion, the MF sender card can send 32 digits to the XCT card. This allows an International Number to be sent in one ANI message, instead of two ANI messages.

ANI signaling

E&M, DX or loop signaling sends ANI information to the Central Office. ANI supports three basic methods: Bell, NT400 and NT500.

- The Bell method interfaces the system to
  - Bell system TOPS, TSPS or CAMA offices
  - Stronger Automatic Toll Ticketing (SATT) systems types 57, 59, 62, and 70A. These systems accept 1+ and 0+ calls from the system using MF pulsing through customer-provided adapter circuits
  - Stromberg Carlson Ticketing Systems
- The NT400 method (Modes A and B) is an interface to the Avaya NT400 ticketing system. Mode A repeats the toll access code (0 or 1) in the called number, whereas Mode B does not.
- The NT500 method (Modes A, B and C) interfaces to Avaya NT500 ticketing systems.
  - Mode A repeats the Access Code (0 or 1) in the called number format for Central Offices that use MF outpulsing and combined trunk groups.
  - Mode B does not repeat the access code.
  - Mode C is used in Central Offices with MF outpulsing and trunk groups dedicated only to 1+ or 0+ calls.

The Bell and the NT400/500 methods have different supervisory signals and different number formatting, as illustrated in Figure 5: Supervisory signals (Bell method) on page 427. Additionally, there are formatting differences between the NT400 and NT500 method. Table 121: Called and calling number information format (Bell method) on page 427 through Table 124: Possible combinations of trunk types and ANI methods on page 430 summarize the possible combinations of trunk types and ANI signaling methods.

The MF sender cable allows the system to independently outpulse up to 16 digits (including starting and ending digits, called KP and ST respectively) in each of the 30 possible network loop time slots. With the E.164/ESN Numbering Plan Expansion feature, the MF sender can send up to 32 digits. Therefore, an International Number can be sent in one ANI message, instead of two ANI messages.

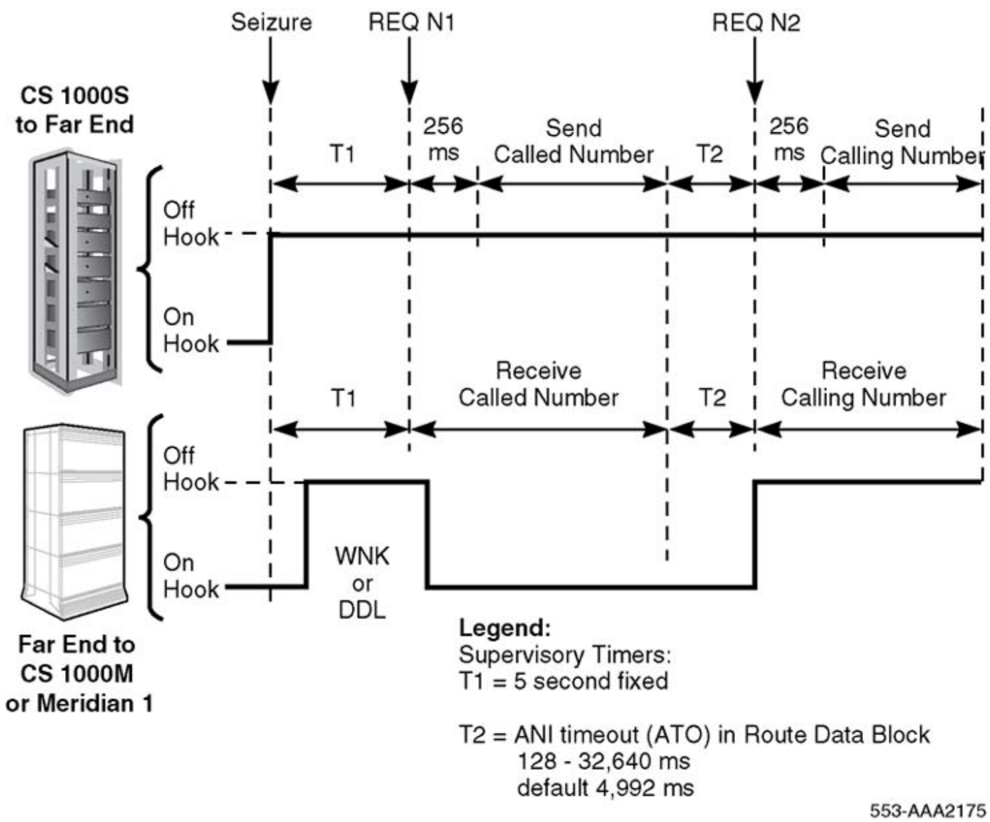


Figure 5: Supervisory signals (Bell method)

Table 121: Called and calling number information format (Bell method)

Legend:
0+ = Operator-assisted call, more digits dialed
0- = Operator-assisted call, no other digits dialed
00+ = Toll operator assisted call, and any other digits dialed
00- = Toll operator assisted call, no other digits dialed
1+ = DDD call
CC = Country code
NN = National number
ID = Information digit
KP = Prepare for digits signal
ST = End of pulsing
STP = Premium
ST2P = Identifier error

Dial Pulse (DP) sending of called numbers			
Call type	Called number	Calling number	
		Regular trunk group	Supervisory trunk group
0	seizure no digits	KP+ID+7D+STP	KP+ID+ST3P
0+7/10D	7/10D	KP+ID+7D+STP	KP+ID+7D+STP
1+7/10D	7/10D	KP+ID+7D+ST	KP+ID+7D+ST2P
011+CC+NN	11+CC+NN	KP+ID+7D+ST	KP+ID+7D+ST2P
01+CC+NN	1+CC+NN	KP+ID+7D+STP	KP+ID+7D+ST3P
010	10	KP+ID+7D+STP	KP+ID+7D+ST3P
Modified Bell Multifrequency sending mode (M2B)			
Call type	Called number		Calling number
	Regular trunk group	Super trunk group	
0	KP+STP	KP+ST3P	KP+ID+7D+ST
0+7/10D	KP+7/10D +STP	KP+7/10D+ST3P	KP+ID+7D+ST
00	KP+0+STP	KP+0+ST3P	KP+ID+7D+ST
00+7/10D	KP+0+7/10D +STP	KP+0+7/10DST3P	KP+ID+7D+ST
1+7/10D	KP+7/10D+ST	KP+7/10D+ST2P	KP+ID+7D+ST
011+CC+NN	KP+1+CC+NN +ST	KP+1+CC+NN +ST2P	KP+ID+7D+ST
01+CC+NN	KP+1+CC+NN +STP	KP+1+CC+NN +ST3P	KP+ID+7D+ST
010	KP+1+STP or KP+10+STP	KP+1+ST3 or KP +10+ST3P	KP+ID+7D+ST or KP +ID+7D+ST

Table 122: Called and calling number information format (NT400 method)

Mode	Call type	Called number	Calling number
A	0+	KP+0+7/10D+ST	KP+CAT+7D+ST[1]
	0-	KP+0+ST	KP+CAT+7D+ST[1]
	1+	KP+1+7/10D+ST	KP+CAT+7D+ST[1]
B	1+	KP+7/10D+ST	KP+CM+CAT+7D+ST[1]

Mode	Call type	Called number	Calling number
	0-	KP+ST	KP+CM+CAT+7D+ST[1]
	1+	KP+7/10D+ST	KP+CM+CAT+7D+ST[1]
Legend:			
CM	=	1 (for 1+ calls)	
	=	STP (for 0± calls)	
CAT	=	XX (category digits)	
X	=	0, 1,...,9, and XX is customer-defined data defining the type of long-distance call	
ST[1]	=	ST (normal)	
	=	ST2P (identifier failure)	
ST[2]	=	ST2P (identifier error)	
	=	KP (station-to-station 1+)	
	=	STP (premium 0±)	

Table 123: Called and calling number information format (NT500 method)

Mode	Call type	Called number		Calling number
		Dial Pulse (DP) sending	Multifrequency sending	
A	0+	0+7/10D	KP+0+7/10D+ST	KP+CAT+7D+ST[1]
	0-	0	KP+0+ST	KP+CAT+7D+ST[1]
	1+	1+7/10D	KP+1+7/10D+ST	KP+CAT+7D+ST[1]
B	0+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[2]
	0-	not applicable	KP+ST	KP+CAT+7D+ST[2]
	1+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[2]
C	0+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[1]
	0-	not applicable	KP+ST	KP+CAT+7D+ST[1]
	1+	not applicable	KP+7/10D+ST	KP+CAT+7D+ST[1]
Legend:				
CM	=	1 (for 1+ calls)		
	=	STP (for 0± calls)		
X	=	0, 1,...,9, and XX is customer-defined data defining the type of long-distance call		

Mode		Call type	Called number: Dial Pulse (DP) sending	Called number: Multifrequency sending	Calling number
ST[1]	=	ST (normal)			
	=	ST2P (identifier failure)			
ST[2]	=	ST2P (identifier error)			
	=	KP (station-to-station 1+)			
	=	STP (premium 0±)			

Table 124: Possible combinations of trunk types and ANI methods

Trunk type	Bell	NT400 A	NT400 B	NT500 A	NT500 B	NT500 C
CAMA-MF	A	A	A	A	A	A
CAMA-DP	A	N	N	A	N	N
CCSA-MF	A	A	A	A	A	A
Legend:						
A = Allowed						
N = Not allowed						

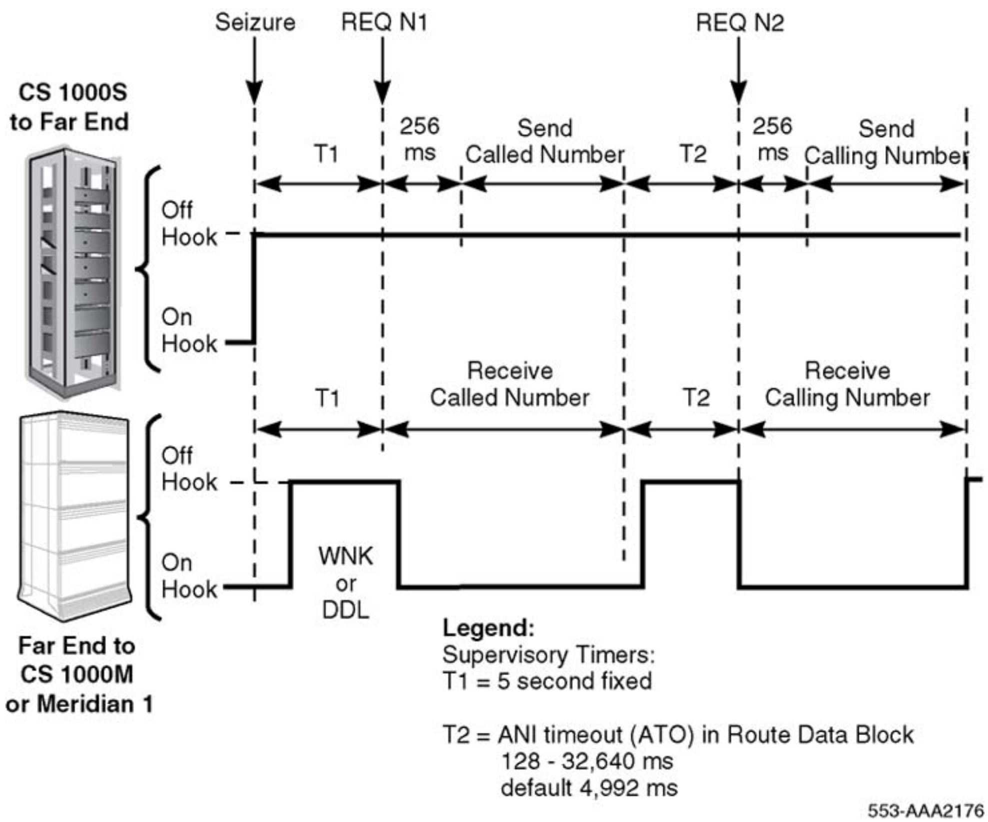


Figure 6: Supervisory signals (NT400/500 method)

Calling and called number information

The called number information always includes the Directory Number (DN) dialed (typically seven or ten digits). The information can also include the toll access code (typically 0 or 1). Multifrequency (MF) sending includes additional control signals such as KP (preparatory digits) or ST (end of pulsing).

The calling number information is always sent in MF. It consists of a calling Directory Number (always seven digits), the preparatory and end-of-pulsing signals and other auxiliary signals. For example, an information digit with the Bell method and class mark and category digits with the NT methods.

Each customer system is assigned a three, four, or five-digit Automatic Number Identification (ANI) Listed Directory Number that identifies the customer to the toll office. The calling number for ANI is obtained by combining the ANI LDN with one of the following:

- Analog (500/2500 type) telephone: Directory Number (DN) of the telephone
- Meridian 1 proprietary telephone: primary DN of the telephone

- Attendant: ANI attendant number specified on a "per customer" basis
- TIE trunk: ANI trunk number specified on a "per trunk group" basis.

The Directory Number Expansion (DNXP) package allows an internal DN to have up to seven digits. If the system is equipped with this package, all DN types listed can be expanded to seven digits maximum. Their combined length with the ANI LDN must remain at seven digits.

The ANI Listed Directory Number is based on the customer's dialing plan. Otherwise, only the leading digits of a DN (station, attendant or TIE trunk) are retained in the ANI calling number. The full seven digits of a DN can be used as the ANI calling number, provided that no ANI Listed Directory Number is configured.

The calling number information is obtained immediately before being sent. Calls that are modified (For example, calls that are attendant extended or transferred) are billed against the party that initiated the trunk call. (This publication is consistent with Automatic Identification of Outward Dialing).

Automatic Number Identification (ANI)/Central Automatic Message Accounting (CAMA) Enhancement

Two call types allow the ANI Bell method to handle 00- and 00+ calls. Customers dialing 00 can transmit KP + 0 + STP to access toll operator assistance. When 0 is dialed, customers can transmit KP + STP to access local operator assistance. Table 125: Actions taken with 00- and 00+ calls on page 432 shows the actions taken by calling 00 and other combinations starting with 0.

Table 125: Actions taken with 00- and 00+ calls

Called number	Bell MF M1A action taken	Bell MF M2B action taken
0	KP + STP	KP + STP
0 + 7/10D	KP + 7/10D + STP	KP + 7/10D + ST3P
00	Overflow	KP + 0 + ST3P
00 + 7/10D	Overflow	KP + 0 + 7/10DST3P

After an ANI/CAMA route has been accessed, the system receive digits representing the called number.

M1A represents the current Bell MF signaling mode. M2B represents the modified Bell MF signaling mode.

Automatic Number Identification/Central Automatic Message Accounting

CAMA routes using Bell MF signaling Mode B outpulse KP + 0 + ..., + START and allow 00- and 00+ calls. 00- and 00+ calls are denied for routes using a different signaling mode.

Controlled Class of Service Allowed (CCSA)

CCSA routes do not support ANI/CAMA.

Route Selection (RS) - Automatic Number Identification

Route Selection for ANI does not support 00- and 00+ dialing. Calls made using 00+ or 00- are treated as 0+ calls. The RS-ANI Data Block determines the 0+ call routing.

Operating parameters

Automatic Number Identification (ANI)/Digital Trunk Interface (DTI) supports CAMA trunks. CCSA-ANI trunks are not supported.

ANI/CAMA operates on a route basis and applies to CAMA routes using the Bell MF signaling method only.

All route members must have a Multifrequency Route (MFR) Class of Service (CLS).

ANI/CAMA is not supported over Dial Pulse trunks. When activating this feature, do not use mixed trunk members.

If 1 or 0 is not dialed following the Trunk Access Code, the system intercepts all outgoing calls over CAMA trunks. This restriction does not apply to outgoing calls over CCSA-ANI trunks.

The two trunk cards mentioned above provide compatibility with the signaling and supervision requirements of CAMA trunks. They also provide a path for the eventual analog transmission of the MF tones and for speech transmission.

Feature interactions

Directory Number Expansion

If the DN Expansion package is equipped, the ANI billing number (ANAT) can have up to seven digits. The total number of digits for ANAT and ANI listed DN (ANLD) cannot exceed seven.

INIT ACD Queue Call Restore

Restored calls do not retain ANI information, unless the call was an incoming call on an M911 trunk.

M911

The Meridian 911 permits special treatment for emergency calls.

Valid Automatic Number Identification combinations

When the system receives a call from a 911 trunk, the trunk receives the ANI information through MF signaling from the Central Office. A valid ANI, received through 911, includes a 1-digit NPD or ID digit followed by a 7-digit calling number. The NPD or identification digit can be displayed directly on the answering telephone display or can be translated to a Numbering Plan Area (NPA) using the Numbering Plan Identification (NPID) translation table in LD 16.

The following are valid ANI digit combinations:

- KP A NXX-XXX ST (where A= the NPD, which can be 0–9);
- KP I NXX XXX ST (where I = an information digit, which can be 0–9;)
- KP I ST (where I = the information digit for ANI failure or Operator Number Identification (ONI). ANI failure is usually designated by a 2 and ONI by a 1); and
- KP A ST (where A denotes maintenance testing, typically the digit 8).

If only one digit is received and that digit is defined in the NPID table as TEST or FAIL, the call is treated as a test case or a call with ANI failure.

Table 126: Interpreting NPD/ID numbers on page 435 shows an example of an NPID table. The last two fields, ANI Failure and Test Calls, are mutually exclusive. If the NPD/ID digit 0 is interpreted as ANI failure, it cannot also be interpreted as a test call.

Table 126: Interpreting NPD/ID numbers

NPI/Info Digit	NPA	ANI Failure	Test Call
0	408	No	No
1	415	No	No
2	NONE	No	No
3	NONE	No	No
4	NONE	No	No
5	NONE	No	No
6	NONE	No	No
7	NONE	Yes	No
8	NONE	No	Yes
9	NONE	No	No

If the NPA is not specified (NPA = NONE), the NPD/ID digit appears on the telephone. Otherwise, the NPA appears on the telephone for calls with a valid ANI.

Seven zeros indicate a failure (for example, MF receive fault, garbled tones or a timeout). After all ANI digits are received or a timeout occurs, the system processes the call.

A test call has no display.

Called and calling numbers are sent from the originator side via MF tones. MF tones are not detected on the terminating side if there is no MFR unit configured. The MFR unit should be configured for 911 trunks. For IPMG system, the MFR unit can only be configured on IPMG card 15 with unit 0 or 1. Changes take effect only after MGC reboot.

There are the following rules for ESA dialed number (ESDN) for CAMA to 911 trunk calls:

- if dialed number length is equal to 1, then this digit should be configured as "1".
- if dialed number length is equal to 2, then these digits should be "11".
- if dialed number length is equal to 3, then these digits should be "911".

Trunk route assignments

The 911 trunk must auto-terminate to a Controlled Directory Number (CDN) defined in LD 23. The start arrangement must be WINK and the Class of Service must be defined as Priority Trunk (APY) and Multifrequency Receiver (MFR).

ANI failure

If ANI information is incorrectly delivered, the call may not have a valid ANI, as indicated by the seven zeros in the display.

ANI failure affects the incoming call's Application Module Link (AML) message, which informs the application with a special DN type value. The 911 caller's DN type Information Element (IE) contains one of these types: ANI with NPD, ANI with ID or ANI failure.

Some Central Offices indicate ANI failure with an 8-digit string consisting of NPD followed by 911-0YYY, where YYY denotes the problem. The ANI string 911-0YYY is not treated as a failure so that the digits appear on the screen rather than being overwritten by seven zeros.

Redundancy and call loss requirements are very precise. If the AML terminal display is unavailable (for example, if the host computer is down), the ANI information still appears on the telephone display.

CDR for 911 ANI calls

If CLID is set to YES in LD 17, 911 ANI information is included in CDR Q records (connection records). CDR records affected are Normal Records, Start/End Records, Authorization Code Records, Connection Records (Q, R, F) and Charge Account Records.

The CDR Q record option is not recommended, since the Meridian 911 application does not need Connection Records and they consume valuable CPU real time. The CDRQ record can nonetheless be configured to include ANI.

Route Selection (RS-ANI)

The optional Route Selection (RS-ANI) is provided with ANI. RS-ANI routes toll calls automatically through specified trunks to toll offices and routes local calls through CO trunks to local switching offices.

To place an outgoing CO call, the station user dials the RS-ANI Access Code (typically 9), followed by a CO Directory Number. If the user dials 0 or 1 after the Access Code, the call routes through a toll trunk group; otherwise, the call routes through a CO trunk group.

Operation

After receiving the RS-ANI Access Code, the system sends the user the second dial tone. The user has 30 seconds to dial a digit or digits. Following this time frame, the system removes the dial tone and provides overflow tone for an additional 15 seconds. The second dial tone is removed after the first digit or digits are dialed. Table 127: RS-ANI operation on page 437 shows the system action that corresponds to the digit dialed.

Although it does provide an overflow tone if the user presses the octothorpe key (#), the system ignores the asterisk (*) key. If 0# is dialed, the system activates a 4-second timer and times out.

Table 127: RS-ANI operation

Digit dialed	System action taken
0	A four second timer starts to monitor the next digit dialed. Routing is based on this digit, as follows:
none	The timer times out and the call (0-) routes through the trunk group specified for 0- calls.
1	The timer cancels, and the call (IDD) routes through the trunk group specified for 1+ of IDD calls.
2–9	The timer cancels, and the call (0+) routes through the trunk group specified for 0+ calls.

Trunk types

TIE trunks access RS-ANI as stations do, but all other trunks are intercepted. Any type of trunk can be used for RS-ANI, with the exception of special-purpose trunks such as Paging, Dictation or Recorded Announcement. Normally, the trunk routes shown in Table 128: Trunk route types on page 437 are used.

Table 128: Trunk route types

Call type	Trunk type
0±	Central Automatic Message Accounting (CAMA)
1+, 011+, 01+, 010-	Central Automatic Message Accounting (CAMA)
other	Central Office (CO)

Class of Service options

Conditionally Unrestricted station Class of Service places non-ARS-handled toll calls through ANI. See *Software Input Output – Administration, NN43001-611* to implement this option. See Table 129: RS-ANI Class of Service options on page 438 for RS-ANI Class of Service options.

Table 129: RS-ANI Class of Service options

Option	Explanation
UNR	Allowed to receive calls from and originate calls to the exchange network (CO, FX, WATS). This includes toll calls.
CUN	UNR for calls placed through ARS and for calls placed through ANI
	TLD for all other calls
CTD	UNR for calls placed through ARS
	TLD for all other calls
TLD	Allowed to receive calls from the exchange network; allowed dial access to local exchange network; allowed access to toll network by means of system attendant only; denied access to exchange operator
Legend	
UNR = Unrestricted CUN = Conditionally Unrestricted CTD = Conditionally Toll Denied TLD = Toll Restricted Service	

New Flexible Code Restriction

Calls from Toll Denied (TLD) stations routed by Automatic Number Identification (ANI) are subject to NFCR. Calls placed by Conditionally Toll Denied (CTD) and Conditionally Unrestricted (CUN) Class of Service stations subject to ANI are treated as unrestricted calls.

Trunk Optimization

ANI trunks allow the Trunk Optimization (TRO) feature to be used whenever calls are routed over PRI and ISL trunks. For more information, see *ISDN Basic Rate Interface Feature Fundamentals* (NN43001-580).

Feature packaging

Automatic Number Identification (ANI) is package 12. The following packages are also required:

- ANI Route Selection (ANIR) package 13, which requires:
  - Automatic Number Identification (ANI) package 12

Feature implementation

Task summary list

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

Table 130: LD 15 - Configure the ANI customer data.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ANI	Automatic Number Identification.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
ANAT	xxx...x	ANI billing number for attendants making ANI calls. (The total number of digits in ANAT and ANLD cannot exceed seven digits.)
ANLD	xxx...x	ANI listed DN for billing purposes (0-5 digits). (The total number of digits in ANAT and ANLD cannot exceed seven digits.)

Table 131: LD 16 - Configure the Centralized Automatic Message Accounting (CAMA) route data.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya CS 1000E system.
TKTP	CAM CAA	SIGL = Bel, NT4, or NT5. SIGL = Bel.
SIGL	BEL NT4 NT5	Bell method signaling. ITT-North NT400 signaling (only if TKTP = CAM). ITT-North NT500 signaling (only if TKTP = CAM).
FORM	M1A M2B M3C	For BEL, NT4, or NT5 (NT4 and NT5 not applicable if TKTP = CAA). For BEL, NT4, or NT5 (NT4 and NT5 not applicable if TKTP = CAA). For NT5 (only if TKTP = CAM).
ICOG	OGT	Outgoing.
ACOD	xxxx	Access Code.
ID	0-9	Identification digit for CAMA routes (for BEL).
CAT	00-99	Category digits for CAMA routes (only if TKTP = CAM). For NT4 and NT5.
STRK	(NO) YES	(Disable) enable super trunk group feature (Bell method signaling only).
SPTO	(NO) YES	7- to 10-digit, or 3-digit outpulsing for ANI calls.
ANKP	(NO) YES	(Do not) suppress KP signal on ANI calls.
CNTL	(NO) YES	(Do not) allow changes to timers.
- TIMR	ATO 128–65,408	ANI timeout timer in milliseconds (default is 4,992).
- ANDT	(NO) YES	(Do not) provide ANI dial tone.

Table 132: LD 14 - Configure the Centralized Automatic Message Accounting (CAMA) trunk data.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	CAM CAA	CAMA trunk. CAMA-ANI trunk (SIGL = BEL in LD 16).
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XUT XEM EXUT	Extended Universal Trunk card. Extended E & M trunk card. Enhanced Extended Universal Trunk.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
SIGL	DPN DAS	Digital Private Network Signaling System Number 1. Digital Access Signaling System Number 2.
SUPN	(NO) YES	Answer and disconnect supervision required.
CLS	aaa	Class of Service options.

Table 133: LD 28 - Configure the Route selection data for ANI calls.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RSA	Route selection for ANI.
RASC	xxxx	RS-ANI access code digits.
0-RT	xxxx	Route access code for 0- calls.
0+RT	xxxx	Route access code for 0+ calls.
1RT	xxxx	Route access code for 1+ or IDDD calls.
CORT	xxxx	Route access code for local calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 63: Automatic Number Identification on DTI

Contents

This section contains information on the following topics:

Feature description

Automatic Number Identification (ANI) on Digital Trunk Interface (DTI) extends the ANI feature to digital Central Office (DCO) and Digital Toll Office (DTO) trunks. In addition, the ANI capability is extended to Primary Rate Access (PRA) trunk routes through the Primary Rate Interface.

For more information, see Automatic Number Identification on page 425.

Operating parameters

The NT817 (all vintages) are required to support this feature.

DTI interfaces externally with a digital trunk carrier facility at the DS-1 rate. MF signals pass across this interface in a digitally encoded format.

Supervisory signaling through DTI is accomplished by A&B bit signaling. A&B bit signaling can emulate E&M or loop signaling.

Address (called number) signaling through DTI can be dial pulse or MF. Immediate start or wink start may be used.

Calling number information signaling is done using the MF signaling method.

This enhancement supports the three basic signaling methods for ANI. These are Bell, NT400, and NT500.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This enhancement is included in Automatic Number Identification (ANI) package 12.

Feature implementation

Table 134: LD 16 - Define Central Office or Toll Office port types.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RBD	Route Data Block.
...		
DTRK	(NO) YES	Digital trunk route.
DGTP	DTI	Digital trunk type.
PTYP	(DCO) DTO	CO or Toll Office port type (default DCO).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 64: Automatic Preselection of Prime Directory Number

Contents

This section contains information on the following topics:

Feature description

Automatic Preselection allows a user to select the Directory Number (DN) assigned to key zero by lifting the handset. It is not necessary to operate the DN key to get dial tone or to answer an incoming call. The DN assigned to key zero is referred to as the Prime Directory Number (PDN) for that telephone.

Operating parameters

The Automatic Preselection feature does not apply to single-line telephones.

Feature interactions

Automatic Redial

If a call is processed on key 0 and the calling party lifts the handset and selects the Prime Directory Number (PDN), this is interpreted as accepting a redialed call.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 135: LD 11 - Assign PDN to key 0 on proprietary telephone.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
...		
CLS	CLS	Class of Service options
	(PDN)	Primary Directory Number
...		
KEY	xx aaa yyyy (cccc or D) zz..z	Telephone function key assignments The following key assignments determine calling options and features available to a telephone. Note that KEY is prompted until just a carriage return <cr> is entered. • xx = key number 0 • aaa = SCR, Single Call Ringing • yyyy = PDN, Primary Directory Number • zz..z = additional information required for the key aaa.

Prompt	Response	Description
		The cccc or D entry deals specifically with the Calling Line identification feature. Where: cccc = CLID table entry of (0)-N, where N = the value entered at the SIZE prompt in LD 15 minus 1. D = the character "D". When the character "D" is entered, the system searches the DN keys from key 0 and up, to find a DN key with CLID table entry. The CLID associated with the found DN key will then be used.

Feature operation

With this feature enabled, lifting the handset automatically selects the DN assigned to key zero to receive dial tone or answer an incoming call on that key.

Chapter 65: Automatic Redial

Contents

This section contains information on the following topics:

Feature description

Automatic Redial (ARDL) extends the redialing capabilities of the Ring Again and Network Ring Again features. The redialing capabilities of this feature reside at the system level. The system generates redialing attempts that allow the calling party to redial a busy public network subscriber using analog or digital trunks.

This feature is applicable when a calling party dials a public network subscriber number and receives a busy indication. Instead of attempting repeated redial efforts, the calling party can activate ARDL by pressing the Ring Again (RGA) key.

Once activated, the ARDL feature requests the system to automatically redial the attempted dialed number until a successful call termination is completed or until the configured number of redial attempts is reached. A successful call termination is determined when one of the following occurs: a tone detector attached to the call detects a ringback tone, an answer signal is received or an ISDN signaling trunk indicates call termination.

When a successful call termination is detected from the far end, the calling party hears the called party through the telephone's loudspeaker. The calling party must accept the redialed call within a specified time limit. If not, the redialed call is dropped and not redialed.

Multi-Automatic Redial permits simultaneous activation of the Automatic Redial feature on several RGA keys. This allows more than one number to be redialed in succession. Each

Automatic Redial call is attempted once and then another number is attempted. Multi-ARDL numbers are dialed in order of activation.

One set of ARDL calls can be associated with one DN key. Another set of ARDL calls can be associated with a different DN key. This option facilitates the use of the ARDL feature by a secretary who works for several managers. Each manager's DN could be on the secretary's telephone. A secretary activates the ARDL feature to call different calling groups on both DNs. After a successful call termination, the accepted call is easily accessed by the appropriate manager.

All ARDL requests are associated with the calling party's DN key. Therefore, when the called party is being redialed the calling party's DN key is busy. If the calling party is busy on another DN, the ARDL attempts are redialed on hold. When a successful call termination is completed, the system alerts the calling party by buzzing the telephone. While ARDL is activated, the calling party's telephone can be used for incoming/outgoing calls.

ARDL can be activated on a call that has originated from a Single Call Ringing (SCR), Single Call Non Ringing (SCN), Multiple Call Ringing (MCR), Multiple Call Non Ringing (MCN), Private Line key or Hot Line key. The ARDL request is associated with the key from which the call was made. If this key is free, the system attempts to dial the number until a successful call termination is detected and provided a free trunk is available.

Operating parameters

The Ring Again feature must be enabled to operate the ARDL feature.

This feature is only supported on Central Office (CO) and TIE trunks.

The ARDL is supported on proprietary telephones, excluding M2317 telephones. It is recommended that telephones be equipped with display, handsfree and loudspeaker. Analog (500/2500) type telephones do not support this feature.

The ARDL feature cannot be activated on data calls.

ARDL can only redial if the Directory Number (DN) key on the calling party's telephone is idle. For this feature application, only a single external number can be stored against the Ring Again key.

Network Ring Again features do not interfere with the ARDL feature. ARDL is only activated after all Network Ring Again attempts have failed. When ARDL is activated, redial attempts continue with the ARDL feature. ARDL does not support the failure of a DPNSS1 call attempt.

The ARDL feature does not impact the operation of the Ring Again feature on internal calls.

The tone detector is not allocated to detect non-busy tones for off network trunks that have on-board busy tone detectors such as an Extended Flexible Central Office Trunk (XFCOT). Only a busy tone is detected. Accordingly, an Automatic Redial call is considered a successful call even though an overflow tone is sent from the far end.

With the exception of trunks that have on-board busy detectors or an end-to-end Integrated Services Digital Network (ISDN) call, a tone detector is required for all ARDL calls.

If a trunk is not equipped with answer supervision, an ARDL call is redialed once only and then the redial request is cancelled.

The busy tone detector capability is limited to the current tone detector hardware.

This feature introduces the following three timers that control the operation of ARDL:

- The Automatic Redial Acceptance Timer is the maximum allotted time that the calling party has to respond to an ARDL call.
- The Automatic Redial Retry Timer controls the time between successive ARDL retries.
- The Tone Detector Response Timer controls the tone detector response and is defined in LD 16.

Feature interactions

Access Restrictions Trunk Group Access Restrictions

The Access Restriction/Trunk Group Access Restrictions of an ARDL redialed call are those restrictions that were applied when the call was initiated. These initial restrictions are not changed.

Attendant Barge-In

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

Attendant Break-In Attendant Busy Verify

Attendant Break-In and Attendant Busy Verify are not permitted on a proprietary telephone that is used for an ARDL call. These restrictions avoid creating a conference when the tone detector is involved in the call.

Attendant Blocking of Directory Number

An ARDL redialed call is blocked from the calling party if an attendant uses the Attendant Blocking of Directory Number feature on the calling party's DN.

Attendant Recall Call Park Call Transfer Conference

No Hold Conference Privacy Release

When an Automatic Redial (ARDL) call is not accepted by the calling party, the following keys are ignored if pressed: Attendant Recall (ARC), Call Park (PRK), Call Transfer (TRN), Conference (A03 or A06), No Hold Conference (NHC) and Privacy Release (PRS).

Autodial

ARDL can be activated on a dialed number using the Autodial (ADL) key.

Automatic Line Selection

Manual Line Selection, Outgoing Line Selection or Prime Line Selection is interpreted as accepting the ARDL by the calling party.

Automatic Preselection of Prime Directory Number

If a call is processed on key 0 and the calling party lifts the handset and selects the Prime Directory Number (PDN), this is interpreted as accepting a redialed call.

Automatic Set Relocation

If the calling party's telephone is relocated, the ARDL request is cancelled.

Call Detail Recording

The calling party's DN is charged even though a call is not accepted. This occurs because the resources are booked for ARDL attempts.

If Call Detail Recording (CDR) is configured on external calls, additional CDR records are produced. This occurs because each redial attempt produces a CDR record.

Calling Party Privacy

The calling party and called party have the same Calling Party Privacy considerations.

Digit Display

Dialed numbers are displayed when the ARDL feature is activated. The calling party can dial digits even though a busy tone indication is given.

Digits dialed while on hold are not displayed. When the calling party accepts a redialed call, the dialed numbers are displayed. If the Display (DSP) key and appropriate RGA key are pressed while a call is on hold, the number redialed is displayed.

Directory Number - Multiple Appearance

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

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

Enhanced Hot Line

An ARDL call can be activated from an Enhanced Hot Line key. However, the call is only redialed when the calling party's HOT key is free.

Last Number Redial

An ARDL call can be activated on a number dialed using the Last Number Redial (LNK) key or by pressing the DN key twice. The ARDL number is saved as the last number redialed.

Line Load Control

ARDL attempts are controlled and restricted by Line Load Control.

Network Alternate Route Selection Network Speed Call

ARDL can be activated on a Network Alternate Route Selection DN or Network Speed Call.

New Flexible Code Restriction

ARDL calls must pass New Flexible Code Restriction (NFCR) checks. If the redialed number is restricted, the ARDL request is cancelled.

Override

An ARDL call cannot be overridden. This is done to avoid creating a conference when a tone detector is involved.

Pretranslation

ARDL can be activated on a number that has passed the Pretranslation process. However, on an ARDL call the Pretranslation process is not used.

Privacy

If the ARDL call is redialed on a number that is shared with any single line telephone, the ARDL call is accepted when the single line telephone goes off-hook.

Privacy Override

When the Privacy Override feature is activated on the MADN key and the one telephone activates ARDL, this call can be accepted by other telephones.

Private Line Service

An ARDL call can be activated on a Private Line Service key. The call can only be redialed when the calling party's PVR or PVN key is free.

R2 Multifrequency Compelled Signaling

A successful ARDL call dialed through a R2 Multifrequency Compelled Signaling (MFC) trunk is determined by the tone detector (TDET) and MFC. If MFC signaling detects that the call has failed, the ARDL call is cancelled in the same manner as a TDET. If R2 MFC does not detect a call failure a TDET is connected to the call as a regular ARDL call.

Scheduled Access Restrictions

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

Speed Call System Speed Call Stored Number Redial

The Automatic Redial (ARDL) feature can be activated on a call using Speed Call (SCL), System Speed Call (SSU/SSC) or Stored Number Redial (RDL) keys.

Speed Call on Private Lines

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

Feature packaging

Automatic Redial (ARDL) requires the following packages:

- Automatic Redial (ARDL) package 304
- Ring Again (RGA) package 1

Outpulsing of Asterisk and Octothorpe (OPAO) package 104 and Automatic Redial (ARDL) package 304 are mutually exclusive. The ARDL package is turned off automatically if both packages are equipped.

Feature implementation

Task summary list

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

Table 136: LD 13 - Define Tone Detector Units

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	TDET	Tone Detector data block.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.

Table 137: LD 15 - Define Automatic Redial

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Change features and options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
...		

Prompt	Response	Description
- ARDL_ATTEMPT	1-(30)-60	Number of Automatic Redial attempts.
REQ:	CHG	Change.
TYPE:	TIM	Change Timers.
CUST	0-99	Customer number For Large Systems
- ARDL_ACCEPT	0-(20)-60	Automatic Redial Acceptance Timer in seconds. Odd number entries are rounded up to the next even number and echoed back with a message.
- ARDL_RETRY	10-(30)-60	Automatic Redial Retry Timer in seconds. Odd number entries are rounded up to the next even number and echoed back with a message.

Table 138: LD 16 - Define Automatic Redial Tone Detector Response Timer

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
...		
CNTL	YES	Changes to controls or timers.
TIMR	RTD 0-(12)-60	Tone Detector Response Timer in seconds. Odd number entries are rounded up to the next even number.

Table 139: LD 87 - Define Automatic Redial Network Route Selection

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
CUST	xx	Customer number, as defined in LD 15
FEAT	NCTL	Network Control Feature.
...		
NCOS	(0) - 99	Network Class of Service group number.

Prompt	Response	Description
- ARDL	(A) I	A = Automatic Redial network route selection allowed from all route sets (initial and extended). I = Automatic Redial network route selection allowed from initial set of routes only.

Table 140: LD 11 - Assign Automatic Redial Class of Service and Key

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
CLS	(RDLA) RDLD	Automatic Redial allowed (default). RDLD = Automatic Redial denied.
KEY	xx RGA	Ring Again key assignment.
KEY	xx RGA	Ring Again key assignment for Multi-Automatic Redial capability.

Chapter 66: Automatic Set Relocation

Contents

This section contains information on the following topics:

Feature description

Automatic Set Relocation (ASR) and Modular Telephone Relocation (MTR) move a telephone to another location without the intervention of a craftsperson. MTR reduces the number of steps required to relocate the Meridian Modular Terminals.

With ASR, Directory Numbers (DNs) and features assigned to the telephone are maintained. Up to 32 telephones can be relocated at any one time. The following access codes are associated with this feature:

- Special Prefix code (SPRE) relocation code 81 SPRE codes are system codes enabling analog (500/2500 type) telephones to utilize additional telephone features. Refer to the Telephones feature module in this guide.
- Flexible Feature Code (FFC) relocation number FFCs are user programmable codes that enable analog (500/2500 type) telephones to access certain telephone features. Refer to the Flexible Feature Code feature module in this guide.
- Security code You must enter the security code before a telephone can be moved.
- Identification code The identification code is user selectable, and can be any four-digit number (excluding the symbols * and #). (MTR does not require this code.)

This feature is also used to install and enable line cards to make unused telephone locations available for telephone relocation. Adding the first telephone on a line card by using the Service

Change overlay enables that card (if it is not already enabled). Removing the last telephone from a line card leaves that card enabled; it does not disable the card.

Automatic Set Relocation (ASR) requires the circuit units on digital line cards used for supplementary power to be specified as power units in LD 12. This allows the system to disable signaling to these units, while leaving unequipped units enabled for telephone relocation. If power units are not specified, they generate erroneous messages and may disable the entire card.

After putting a telephone back into service, the craftsperson should wait at least 20 seconds before using the telephone.

Modular Telephone Relocation

Modular Telephone Relocation enhances ASR to make relocating Meridian Modular Telephones simpler and faster (by omitting the requirement for an identification code). The following telephones support Modular Terminal Relocation:

- M2006
- M2008
- M2016S
- M2216
- M2616

When a telephone is relocated out, a relocation block is automatically built to store the relocation information in the protected data area. The relocation block includes the old Terminal Number (TN), the terminal ID information, the serial number of the telephone, and feature information. If a data dump occurs, the relocation block is not copied to the disk.

Modular Telephone Relocation uses the unique serial number and terminal ID of the Meridian Modular Telephones (instead of the identification code) to identify the one being relocated. This reduces the number of steps needed for relocation.

A telephone's successful relocation is indicated by a 180-millisecond buzz through the telephone's loudspeaker, not a tone through the handset. The buzz occurs after the telephone is plugged into the new location, and the parameter download to the Meridian Modular Terminal is complete.

Modify the relocation table

The relocation table contains information regarding the telephone's serial number, Terminal Number, and terminal identification. When a telephone is relocated OUT, the table maintains the necessary telephone information. When the telephone is relocated IN, the system searches the table for that telephone's information. When the information is found, the data is moved to the new location. The telephone data is then removed from the relocation table.

Through LD 50, the serial number or any terminal ID information may be modified while the telephone is relocated out (before it has been relocated back in). For example, use LD 50 when replacing a telephone with another one of the same type with a different serial number or terminal ID, but the same key configuration.

LD 21 prints information about telephones that have been relocated out.

The IDU (ID for Unit) command in LD 32 determines the telephone's serial number and ID information.

Operating parameters

A single-line telephone must be relocated to a vacant position on an analog (500/2500 type) Line Card.

A digital telephone must be relocated to a vacant position on a Digital Line Card (DLC) or Integrated Services Digital Line Card (ISDLC) in the switch.

Moving a telephone from an off-premise to on-premise location or vice versa is not recommended, as incorrect pad values on connections may result.

A Manual Line telephone cannot be relocated using the Automatic Set Relocation feature.

The relocation table allows a maximum of 32 telephones to be relocated out at one time.

A relocated out telephone cannot be relocated in to an already defined TN. A telephone being relocated in must be plugged into a TN location that currently has no assigned telephone information.

Automatic Call Distribution (ACD) agent telephones with an associated supervisor and the ACD supervisor telephones cannot be relocated.

If a data dump occurs while a telephone is relocated out, a SYSLOAD returns the telephone to its original TN location. If a telephone was in the relocated out state when the last data dump occurred, and has since relocated in, another data dump is necessary. The second data dump prevents a SYSLOAD from returning the telephone to its previous TN location.

When Modular Telephone Relocation is used and the overflow tone is returned during relocation out, the relocation attempt is abandoned. Try the relocation again.

When Modular Telephone Relocation is used, there is a slight delay between the time the telephone is plugged in and the buzz. The buzz occurs after the telephone is relocated in, enabled, and downloaded. This delay is traffic dependent. If no buzz is received, the relocation is unsuccessful.

When Modular Telephone Relocation is used and a telephone is relocated out, a Customer Service Change (CSC) message containing the old TN number, serial number, and terminal ID is displayed on the TTY. When a telephone is relocated in, a CSC message containing the old TN and new TN is displayed. These messages are placed in the History File.

When Modular Telephone Relocation is used and a SYSLOAD occurs before a data dump completes, the data for all telephones relocated in or out is lost. Return the telephones to their original location and repeat the relocation process.

Feature interactions

Automatic Redial

If the calling party's telephone is relocated, the Automatic Redial request is cancelled.

Busy Forward Status

BFS is not compatible with Automatic Set Relocation.

Call Forward No Answer Hunting

Calls will not hunt or forward no answer to a telephone that is being relocated.

Call Forward Ring Again

If Call Forward, or Ring Again is active when a telephone is relocated, the feature is deactivated.

China - Flexible Feature Codes - Busy Number Redial

Enhanced Flexible Feature Codes - Busy Number Redial

Busy Number Redial is deactivated when a telephone is relocated.

Hunting

Calls will not hunt to a telephone that is being relocated

Make Set Busy

If Make Set Busy is active when the telephone is relocated, Make Set Busy remains active.

Multiple Appearance DN Redirection Prime

The original Multiple Appearance Directory Number Redirection Prime (MARP) TN is restored when the telephone relocates.

When Automatic Set Relocation or Meridian Modular Terminal is used to move a telephone, the telephone's MARP designations are maintained. If the TN is a MARP for one or more DNs, the system maintains the MARP TN. A system message indicates the telephone relocation.

When a telephone leaves the system due to Automatic Set Relocation, the following CSC message appears:

CSC010 x y

x = old TN (l s c u) for the telephone

y = ID code entered

While the telephone is being relocated, a temporary MARP TN is assigned. The following SCH message appears for each DN associated to the removed MARP TN.

SCH5524 DN nnnn NEW MARP l s c u

nnnn = the DN associated with the MARP TN

l s c u = the new default MARP for DN nnnn

The same message given through Attendant Administration displays on the attendant console when a MARP is assigned for a DN. The History File can be configured to store these messages until a printout is requested.

When a telephone reenters the system, the following message appears:

CSC011 x y

x = old TN (l s c u) for the telephone

y = new TN (l s c u) for the telephone

The following message appears again for each changed TN:

SCH5524 DN nnnn NEW MARP l s c u

nnnn = the DN associated with the MARP TN

l s c u = the new MARP TN assigned to DN nnnn

Night Key for Direct Inward Dialing Digit Manipulation

Delete the DRC key from a telephone before performing Automatic Set Relocation. If this is not done, the DRC lamp is activated on the wrong telephone.

Power Fail Transfer

Since Power Fail Transfer is hardwired to certain Terminal Numbers, this feature is not maintained by a telephone when it is relocated.

Feature packaging

Automatic Set Relocation (ASR) package 53 has no feature package dependencies.

Modular Telephone Relocation requires the following:

- Automatic Set Relocation (ASR) package 53
- Meridian Modular Terminals (ARIE) package 170
- Digital telephones (DSET) package 88

Feature implementation

Task summary list

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

1. Table 141: LD 15 - Assign the Automatic Set Relocation security code. on page 467
2. Table 142: LD 10 - Enable/disable line circuits for Automatic Set Relocation. on page 467

3. Table 143: LD 11 - Enable/disable line circuits for Automatic Set Relocation. on page 467
4. Table 144: LD 12 - Gather data for each line circuit to be used as a supplementary power source. on page 468
5. Table 145: LD 17 - Allow ASR messages to be printed at a system terminal or stored in the History File. on page 468
6. Table 146: LD 17 - Allow Automatic Set Relocation messages to be printed at a system terminal or stored in the History File. on page 468
7. Table 147: LD 32 - Query information on page 469
8. Table 148: LD 50 - Remove an entry in the relocation table. on page 469
9. Table 149: LD 21 - Print information in the relocation table. on page 469

Table 141: LD 15 - Assign the Automatic Set Relocation security code.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- SRCD	xxxx <CR> X	Automatic Set Relocation security code; default is 0000; X removes security code.

Table 142: LD 10 - Enable/disable line circuits for Automatic Set Relocation.

Prompt	Response	Description
REQ:	NEW OUT	Configure Automatic Set Relocation. Remove Automatic Set Relocation.
TYPE:	cardslt	500/2500 line circuit for Automatic Set Relocation.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Table 143: LD 11 - Enable/disable line circuits for Automatic Set Relocation.

Prompt	Response	Description
REQ:	NEW	Add new data.
TYPE:	cardmlt	Digital line circuit for Automatic Set Relocation.

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

Table 144: LD 12 - Gather data for each line circuit to be used as a supplementary power source.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PWR	Digital line circuit for supplementary power.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.

Table 145: LD 17 - Allow ASR messages to be printed at a system terminal or stored in the History File.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
IOTB	(NO) YES	(Do not) change input/output terminals or devices.
HIST	(0)-65534	History File buffer length.
- ADAN	NEW CHG aaa x	System terminal device number for Automatic Set Relocation messages. aaa and x = HST. PRT 0-15. TTY 0-15.
- USER	CSC	Customer service change (Automatic Set Relocation) messages.

Table 146: LD 17 - Allow Automatic Set Relocation messages to be printed at a system terminal or stored in the History File.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW CHG aaa x	System terminal device number for Automatic Set Relocation messages. aaa and x = HST. PRT 0-15. TTY 0-15.
- CTYP	aaaa	Card type, where: aaaa = DCHI, MSDL, MSPS, SDI, SDI2, SDI4, or XSDI.

Prompt	Response	Description
- DNUM	(0-15)	Device number printed automatically (same as ADAN number).
- USER	CSC	Customer service change (Automatic Set Relocation) messages.
CUST	xx	Customer number, as defined in LD 15

Table 147: LD 32 - Query information

IDU l s c u		Prints telephone's information.
Use this command to query information about your terminal type, NT code, color, release number, and unique serial number. This command works only for Meridian Modular Terminals.		

Table 148: LD 50 - Remove an entry in the relocation table.

Prompt	Response	Description
REQ	OUT CHG	Remove or change an entry in the relocation table.
TYPE	MTRT	Modular Telephone Relocation Table.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
SER	xxxxxx	Serial number (prompted for changes only).
NTCD	xxxxxxx	NT code (for changes only).
COLR	xx	Color (prompted for changes only).
RLS	xx	Release (prompted for changes only).

Table 149: LD 21 - Print information in the relocation table.

Prompt	Response	Description
REQ:	PRT	Print.
TYPE:	SRDT	Set relocation data.

Feature operation

To use Automatic Set Relocation:

1. Lift the handset.
2. Enter the relocation code (either SPRE 81 or the Flexible Feature Code).
3. Enter the security code. The default is 0000.
4. Enter the four-digit code to identify your telephone. A tone confirms the telephone is ready to be moved.
5. Unplug the telephone and install it at the new location.
6. Wait 30 seconds after plugging the telephone into the new location, lift the handset, and dial the four-digit identifier. A tone confirms the telephone has been moved successfully.

Modular Telephone Relocation

To relocate a telephone using Modular Telephone Relocation:

1. Lift the handset or activate handsfree.
2. Enter the relocation code (either SPRE 81 or the Flexible Feature Code).
3. Enter the security code. The default is 0000.
4. A two-second tone burst confirms that the telephone is relocated out.
5. Unplug the telephone and install it at the new location.
6. The confirmation buzz through the telephone's loudspeaker indicates the telephone is in service.

All calls associated with the telephone receive force disconnect while it is relocated out. The telephone information automatically moves to the relocation table.

Chapter 67: Automatic Timed Reminders

Contents

This section contains information on the following topics:

Feature description

Automatic Timed Reminders alert the attendant when a call extended to a station by the attendant console has not been answered within a predefined period of time. Recall timers for different conditions can be specified by the customer as follows:

- Slow Answer (set in increments of six seconds)
- Camp-On (set in increments of two seconds)
- Call Waiting (set in increments of two seconds)

If no entry is made, the default is 30 seconds in each case.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Overflow Position

After an attendant call has been rerouted using the Attendant Overflow Position feature, there is no automatic timed recall to the attendant or any other DN.

Call Forward by Call Type

Calls eligible for Flexible Call Forward No Answer treatment, and handled by Call Forward by Call Type, use the Call Forward No Answer timer in the Customer Data Block as the recall timer for attendant extended calls. Irrespective of the relative timeout for Automatic Timed Recall, the ringing continues as long as allowed by the Call Forward No Answer Timer.

Call Forward No Answer Call Forward No Answer Second Level

When Call Forward No Answer is activated on a telephone, the slow answer timer begins only after the call reaches its final destination.

Call Park

A Call Park recall to an attendant appears on the Recall Incoming Call Indicator.

Call Waiting Redirection

When Call Forward No Answer (CFNA) is active, the Slow Answer Recall timer begins only after the call reaches its final destination. CFNA has precedence over Attendant Recall for attendant-extended calls. Irrespective of the relative time-out intervals for each feature, ringing continues as long as allowed by CFNA for telephones with CFNA enabled.

Since the Call Waiting Redirection feature applies CFNA treatment to a Call Waiting call, the Call Waiting Redirection feature also takes precedence over the Call Waiting recall timer.

Directory Number Delayed Ringing (DNDR)

If a dialed telephone has DNDR defined, and an attendant re-extends a call without releasing it, the DNDR timing is not reset. If the value of the recall timer is less than that of the DNDR timer, the call is recalled to the attendant before audible notification begins.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 150: LD 15 - Define Recall timers and add/change a Recall Incoming Call Indicator key on attendant consoles.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	TIM	Timers.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- RTIM	xxxx yyyy zzzz	Recall timers. xxxx = slow answer, 0-3,066, in six-second increments (default 30 seconds). yyyy = Camp-on, 0-1,022, in two-second increments (default 30 seconds). zzzz = Call Waiting, 0-1,022, in two-second (increments (default 30 seconds).
TYPE	ATT_DATA	Attendant console options.
- ICI	0-19 RLL	Add RECALL ICI to all consoles.

Feature operation

One optional Recall Incoming Call Indicator (ICI) key is provided on the attendant console for operator-extended recalls.

Chapter 68: Automatic Wake Up

Contents

This section contains information on the following topics:

Feature description

Automatic Wake Up (AWU) provides an efficient wake up service for hospitality and health care environments. It relieves the attendant from having to make wake up calls by providing this service automatically. At the requested time, the system automatically rings the room or extension and connects the called party upon answer to music followed by a recorded wake up announcement.

If the wake up call is answered within a customer-specified number of rings (two to five rings), the system recognizes a completed call and presents the predefined wake up treatment. The system disconnects the AWU call when the called party releases, or when the recording cycle is completed.

The wake up message runs continuously. Upon answering a wake up call, the called party hears music until the message begins again. If the message is 15 seconds long, and the wake up call is answered on the 14th second of the message, the calling party hears one second of music before the message. If the call is answered on the third second of the message, the calling party hears 12 seconds of music first.

The system allows for an alternate recording that can be used for evening wake up calls or when the primary recording is being updated. The secondary recording can also replace the primary recording at a customer-specified time period.

Answer the wake up call

The Wake Up indicator goes dark after the guest answers the wake up call. Customers can set the attendant recall option if the call is unanswered after a specified number of tries (from one to three).

Answering the wake up call for multiple appearance DN telephones is similar to single appearance DN telephones: after the call is answered, the Wake Up indicator goes dark.

The system balances the wake up load over five-minute intervals, generating a maximum of 100 wake up calls per five-minute period. The system processes one wake up call every two seconds during peak periods, and one wake up call every four seconds during lighter periods. A light load is defined as anything less than 60 wake up call requests per five-minute interval.

A wake up request is rejected by the system under the following conditions:

- The wake up request (in units of five-minute intervals) is less than one interval ahead of the current time interval (see Note below).
- The wake up request (in units of five-minute intervals) is less than five intervals before the current time interval. In other words, the wake up request is more than 23 hours and eight intervals in advance.
- The interval requested contains 500 calls already.

Important:

The time interval = (hour x 12) + (minute / 5). Always round down to the nearest five-minute interval.

If the interval requested for a wake-up call already contains the maximum number of calls, the system searches for the next available time interval in the following sequence:

- the five-minute interval before the requested time
- the five-minute interval after the requested time
- the next available five-minute interval within three hours before the requested time

You can also use a Background Terminal (BGD) to enter Automatic Wake Up information. The Background Terminal lets you monitor system operation. One or more terminals can be assigned to access AWU data. You can have data displayed or printed at a preselected time of day.

500 Wake Up calls

The number of Automatic Wake Up calls available per five-minute period is 500 calls.

You can define the number of rings for the call from two to five. If there is no answer after the specified number of rings, the AWU call overflows to the next five-minute interval. The system tries three times to terminate the call before it is recalled to the attendant. You can define the number of wake up attempts, from one to three.

No more than 25 analog (500/2500 type) telephones should be ringing at any one time. To ensure this, set the Number of Rings for Wake Up (NRWU) prompt in LD 15 according to the recommendations listed in Table 151: Recommended number of rings per Automatic Wake Up call on page 477. The NRWU is two to five, with a default of five.

Table 151: Recommended number of rings per Automatic Wake Up call

Time on (seconds)	Time off (seconds)	Maximum number of rings
2*	4*	5*
3	3	2
2	1	5
1	2	5
* North American standards.		

Only 500 AWU calls can be defined for the system, but up to 750 calls can actually be placed. Up to half of the programmed AWU calls unanswered can be carried over to the next five-minute interval. The carry-over from one block to the next is important in limiting the number of calls in the original programmed interval.

For a complete description on programming AWU with the Background Terminal, see *Hospitality Features Fundamentals, NN43001-553*.

Guest Entry of Auto Wake Up (GEWU) Calls

GEWU provides entry of a wake up call from a room telephone. By using the Wake Up key (WUK) on the telephone, guests can program, query (with display), or cancel their own wake up calls based on a 24-hour time format.

Requests must be made on a daily basis since the wake up time is automatically canceled after each use.

GEWU does not alter the operation of AWU, but adds a new option to AWU programming. Unless otherwise specified, operating GEWU is the same regardless of whether the telephone has a display. The distinction is that with a display, guests can check their wake up call requests. A dash (–) indicates that no time has been programmed. In addition, when programming a wake up call, the system will search for and display the next available time if the time interval chosen for the wake up call is full. Without a display, the guest can still program and cancel a wake up call.

For Multiple Appearance DN telephones, the wake up time for secondary DNs cannot be queried.

Multi-Language Wake Up (MLWU) Calls

MLWU provides Automatic Wake Up calls in any of up to six languages. You can use any language as long as you have a recording of it available on a Recorded Announcement (RAN) trunk.

At check-in, each guest can choose the language for wake up calls. If no language is assigned, the default language, Language 0, is used.

You can assign a language to a room's telephone at any time by using the Background Terminal (BGD) or Property Management System (PMS). A room DN is valid if it has at least one appearance as a Prime DN (key 0) on a telephone and Controlled Class of Service Allowed (CCSA). Multiple appearance telephones with the same Prime DN may be assigned different languages through Service Change.

You can also assign the language on a TN basis, allowing the language option to be employed outside the hospitality industry without requiring a BGD terminal or the PMS. Refer to LD 10 and LD 11 in *ISDN Basic Rate Interface Installation and Commissioning, NN43001-318* for the "LANG" prompt.

The language remains unchanged until the next language assignment. An AWU language cannot be changed on a call-by-call basis. The customer may, however, optionally clear the language either at check-in or check-out times, using the Background Terminal.

If Automatic Wake Up is enabled, up to six pairs of language-specific RAN routes (both a.m. and p.m. for each language), called Automatic Wake Up routes (AWR), can be configured. The languages, 0-5, correspond to the AWR routes RAN1/RAN2 (for Language 0), LA11/LA12 (for Language 1), up to LA51/LA52 (for Language 5) in the Customer Data Block (LD 15). The only requirement is that the default language routes RAN1 and RAN2 for Language 0 must be defined. If a specific language AWR is not accessible at wake up time, the corresponding primary or secondary default language routes (RAN1 and RAN2) are used.

On a Background Terminal, a customer can define a two-character language identifier to reference the languages. For example, the customer may define Language 0 as EN (English), Language 1 as SP (Spanish), and Language 2 as GR (German). For details on implementing BGD terminal commands, refer to *Hospitality Features Fundamentals, NN43001-553*.

Unanswered Automatic Wake Up calls recall to the attendant if the attendant recall option is on. Upon a recall, the room's language identifier is displayed after the Call Party Name Display (CPND) fields on the attendant console.

Multiple Wake Up Flexible Feature Codes

Multiple Wake Up allows up to four wake-up calls to be entered using a Flexible Feature Code (FFC), and allows those calls to be repeated daily, if desired, by entering a separate FFC. The time is in a four-digit 24-hour format (H1 H2 M1 M2). To activate Repeat Multiple Wake Up, the user dials "MWRA H1 H2 M1 M2".

If a wake-up time has already been entered using the standard Automatic Wake Up Activate (AWUA) FFC, only three other multiple wake-up times may be entered.

To deactivate a single wake-up time, the user enters "MWUD H1 H2 M1 M2", where MWUD is the Multiple Wake Up Deactivate FFC. To deactivate all wake-up times, the user enters "MWUD#". The general Deactivate (DEAF) FFC does not apply to Multiple Wake Up.

If the MWUD FFC is entered again after all wake-up times have been deactivated, confirmation tone is given. If the MWUD FFC is entered again to deactivate a wake-up time that has been already deactivated, overflow tone is given. If an attempt is made to enter an existing wake-up time, confirmation tone is given. If an attempt is made to enter an existing wake-up time as a repeat wake-up time, then that time is activated as a repeat wake-up time. If an attempt is made to enter an existing repeat wake-up time as a single wake-up time, then that time is activated as a single wake-up time. In both cases, confirmation tone is given.

To verify a Multiple Wake Up time, the user dials "AWUV H1 H2 M1 M2" (where AWUV is the existing Verify Automatic Wake Up FFC).

Operating parameters

To operate AWU, a system must have a Background Terminal or attendant console with AWU key, room telephones with Controlled Class of Service Allowed (CCSA), and Recorded Announcement (RAN) trunks.

This feature requires a Background Terminal (BGD).

Each Automatic Wake route requires a minimum of two trunks.

The following hardware is required for the AWU feature:

- NT8D14AH universal trunk card
- a continuous announcement (RAN) machine, such as the Audichron HQ-1 112

A dedicated conference loop is no longer required for the network-enhanced machines.

For the call to utilize both music and a wake up announcement, an AWR route must be installed and the route must be programmed at the RANF prompt in LD 15. The music source can be wired into the audio pairs of the RAN trunk, or music can be recorded on the RAN device.

Automatic Wake Up is only allowed on a telephone's Prime Directory Number (PDN). For telephones in a multiple-appearance arrangement, all telephones are rung; however, only one wake up time can be assigned against the PDN. The system tries the wake up call a customer-defined number of times (from one to three), and then treats it as any other unanswered wake up call. In a single-call arrangement, if any appearance of the DN is busy when the wake up call is made, the wake up call is not presented. In a multiple-call arrangement, the wake up call is presented to all idle appearances.

A wake up key cannot be configured on a data station (a telephone with DTA Class of Service).

There can only be one wake up key per telephone.

Only attendant consoles can have an AWU key. The AWU time to be programmed on digital telephones (using GEWU and a Wake Up key).

Automatic Wake Up and Centralized Attendant Services (CAS) are mutually exclusive.

If the wake up call goes unanswered, or the guest hangs up before the AWU two-second hold time, the system tries the wake up call again in the next five-minute interval. If Attendant recall is enabled, the call transfers to the attendant following the last unsuccessful wake up call attempt.

Maintenance technicians can access any AWU RAN trunk or music trunk with the RAN trunk access code.

For Multiple Wake Up, the FFCs selected must be unique numbers up to seven digits long. They cannot conflict with any DN already in the dialing plan.

The following are not supported for Multiple Wake Up:

- The attendant query for the Multiple Wake Up time
- Multiple Wake Up from attendant administration
- The Background Terminal, Background Terminal Display for Multiple Wake Up
- Traffic for Multiple Wake Up

The Deactivate (DEAF) FFC is not supported for Multiple Wake Up.

Multiple Wake Up is supported only on analog (500/2500 type) telephones.

The Automatic Wake Up feature can be active at the same time as Multiple Wake Up.

If one Automatic Wake Up time has been set using the Automatic Wake Up Activate (AWUA) FFC, only three more Multiple Wake Up calls can be entered using the MWUA FFC.

Feature interactions

Attendant Administration

The Attendant Administration feature does not support data entry or changes for the AWU feature.

Attendant Overflow Position

AWU recalls are not redirected to a customer-defined Attendant Overflow Position DN. Failed wake up calls stay in the attendant queue or ring indefinitely on the console.

Call Party Name Display

All display information associated with Automatic Wake Up (AWU) programming is directed to line three of the display. Names are appended to DNs appearing on line three if they are different from those on line two, or if no DN appears on line two. There is no DN information on line two if the attendant has initiated the AWU process while not on an active call. No DES information is appended, since AWU operates on a DN basis.

Coordinated Dialing Plan

AWU supports Coordinated Dialing Plan as long as an internal DN is used.

Directory Number Delayed Ringing

The Directory Number Delayed Ringing feature is not supported.

Do Not Disturb

When a telephone is configured for Do Not Disturb, a wake up call can still be presented.

Flexible Feature Codes Enhancement

Telephones can activate Automatic Wake Up (AWU) features for their own station with Common Controlled Switching Arrangement Class of Service.

The Automatic Wake Up feature can be active at the same time as Multiple Wake Up.

The attendant query function is not supported for Multiple Wake Up.

Multiple Wake Up from attendant consoles is not supported.

The Background Terminal (BGT) is not supported for Multiple Wake Up.

If one Automatic Wake Up time has been set using the Automatic Wake Up Activate (AWUA) FFC, only three additional Multiple Wake Up calls can be entered using the Multiple Wake Up Activate (MWUA) FFC.

Intercept Computer Dial from Directory - Post-dial Operation

This feature can be requested as follows:

- Press the Wake-up key on the attendant console.
- Dial a DN from the Intercept Computer.

Dial an octothorpe sign "#", and terminate by dialing the requested wake-up time from the attendant console

Manual Line Service

Manual Line or Private Line Services

AWU does not support these features; an AWU call cannot be programmed against a manual line or private line DN.

Multiple Appearance DN

All Multiple Appearance DNs are rung, including both primary and secondary DNs. Programming the wake up request using the Wake Up key applies only to telephones with the primary DN on key 0, and the Wake Up indicator operates as described only on the telephone that is currently programming the wake up request.

In addition, if two or more Multiple Appearance Primary DN telephones program a wake up request at the same time, the last telephone to finish overrides. In other words, all telephones with the same primary DN get the same request time of the last telephone to program a request. If the last telephone cancels the request, all requests are canceled.

When the wake up programming sequence is finished, all Wake Up indicators on Multiple Appearance Prime DNs are updated unless a telephone is in the middle of Wake Up programming.

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

Night Service

Unanswered AWU calls going through Attendant Recall are discarded if the attendant console is in the Night Service mode. AWU may still be programmed when the attendant console is in Night Service.

Pretranslation

When the Pretranslation feature is equipped with AWU, the actual DN, not the pretranslation DN, should be used when programming the AWU call request.

Room Status

Room Status and Automatic Wake Up both use the Background Terminal (BGD). If the WAKE option is selected for the check-in/check-out operation, the wake-up call for that room is canceled after a check-in or check-out operation.

When a guest checks in or out, the room status changes. If an AWU request is still active, it is canceled if it is included as part of the Check In/Out option.

Feature packaging

Automatic Wake Up (AWU) package 102 requires:

- Recorded Announcement (RAN) package 7
- Controlled Class of Service (CCOS) package 81
- Background Terminal Facility (BGD) package 99

Guest Entry of Auto Wake Up is included as part of Automatic Wake Up (AWU) package 102.

Multi-Language Wake Up (MLWU) package 206 requires Automatic Wake Up (AWU) package 102.

Multiple Wake Up FFCs require Flexible Feature Codes (FFC) package number 139.

Feature implementation

Task summary list

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

Table 152: LD 16 - Define the RANF route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E system.
TKTP	AWR	AWU RAN route.

Prompt	Response	Description
RTYP	AUD	Audichron recorder.
- GRD	PLAY IDLE	Ground Start Arrangement where: PLAY = RAN machine sends a ground signal when playing. IDLE = RAN machine sends a ground signal when idle. If the United Kingdom (UK) package 190 is equipped the default response is PLAY, if this package is not equipped the default response is IDLE.
ACOD	xxxx	Trunk route access code. Must be different from RANF ACOD.

Table 153: LD 16 - Define the RAN1 route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya CS 1000E system.
TKTP	AWR	AWU RAN route.
RTYP	AUD	Audichron recorder.
- GRD	PLAY IDLE	Ground Start Arrangement where: PLAY = RAN machine sends a ground signal when playing. IDLE = RAN machine sends a ground signal when idle. If the United Kingdom (UK) package 190 is equipped the default response is PLAY, if this package is not equipped the default response is IDLE.
ACOD	xxxx	Trunk route access code. Must be different from RANF and RAN1 ACODs.

Table 154: LD 16 - Define the RAN2 route.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	RDB	Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number. Must be different from RANF and RAN1.

Prompt	Response	Description
	0-511	Range for Large System and CS 1000E system.
TKTP	AWR	AWU RAN route.
RTYP	AUD	Audichron recorder.
- GRD	PLAY IDLE	Ground Start Arrangement where: PLAY = RAN machine sends a ground signal when playing. IDLE = RAN machine sends a ground signal when idle. If the United Kingdom (UK) package 190 is equipped the default response is PLAY, if this package is not equipped the default response is IDLE.
ACOD	xxxx	Trunk route access code.

Table 155: LD 14 - Define the trunk for RANF.

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

Table 156: LD 14 - Define the trunk for RAN1.

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

Table 157: LD 14 - Define the trunk for RAN2.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	AWR	AWU RAN trunk.
TN	l s c u	Terminal Number. Must be a different TN from RANF and RAN1. Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member number. Must be different from RANF and RAN1.
	0-511 1-4000	Range for Large System and CS 1000E system.

Table 158: LD 15 - Enable Automatic Wake Up in Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	AWU	Automatic Wake Up options.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
- AWU	YES	Activate AWU for a customer.
- ATRC	(NO) YES	(Deny) allow attendant recall.
CONF	0-159	Conference loop number.
- RANF	0-511	Music RAN route number.
- RAN1	0-511	Primary AWR route number.
- RAN2	0-511 <CR>	Secondary AWR route number.
- LA11	X 0-511	Language 1, RAN route 1. X = remove language RAN route definition.
- LA12	0-511	Language 1, AWR route 2.
- LA21	0-511	Language 2, AWR route 1.
- LA22	0-511	Language 2, AWR route 2.
- LA31	0-511	Language 3, AWR route 1.
- LA32	0-511	Language 3, AWR route 2.
- LA41	0-511	Language 4, AWR route 1.

Prompt	Response	Description
- LA42	0-511	Language 4, AWR route 2.
- LA51	0-511	Language 5, AWR route 1.
- LA52	0-511	Language 5, AWR route 2.
- R2BN	hhmm	RAN2 start time.
- R2ED	hhmm	RAN2 end time.
- NRWU	2-(5)	Number of rings for a wake up call
- TAWU	1-(3)	Number of wake up tries for an unanswered AWU call
AWR route number ranges from 0-511 apply to Large Systems only. Range is 0-127 for all other options. Enter "X" to remove a route.		

Table 159: LD 10 - Set language and CCOS for analog (500/2500 type) telephones (on a per TN basis).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500 2500	Telephone type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
LANG	(0)-5	Language number. To remove entry, precede with X.
CLS	CCSA	Controlled Class of Service allowed.

Table 160: LD 11 - Set language and CCOS for Meridian 1 proprietary telephones (on a per TN basis).

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CLS	CCSA	Allow Controlled Class of Service.
LANG	(0)-5	Language number. To remove entry, precede with X.
KEY	xx WUK	Assign a wake up key on a telephone. Must be a key/lamp pair.
To assign a language on a per DN basis, use a Background Terminal.		

Table 161: LD 12 - Allow access to AWU from attendant consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx AWU	Add an AWU key.

Automatic Wake Up Diagnostic:

To check the availability of the delivery of AWR messages, the technician dials the Access Code (ACOD) from a maintenance telephone only. A maintenance telephone is equipped with a MTA Class of Service. Trunks can also be diagnosed in LD 36 by entering the command AWR C R (Test Automatic Wake Up devices associated with Customer (C) and Route (R).

Feature operation

From a telephone with a Wake Up key

To program a wake up call from an idle telephone, follow these steps:

1. Press **Wake Up**. The indicator flashes.
2. Dial the wake up request time, in 24-hour format (7:30 a.m. as 730, 7:30 p.m. as 1930). Telephones with display show a dash followed by the time. If no time is set, a single dash is shown.The indicator keeps flashing.
3. Press **Wake Up**. The indicator goes on steady.

Press the **Release** (Rls) or **PDN** key while programming a wake up request to abort the wake up request. Any previously defined wake up time will remain.

Display telephones If the time interval chosen for the wake up call is full, the system searches for and displays the next available time. If the system cannot find another time, the display shows four dashes (– – – –), and the Wake Up indicator remains flashing. If the system finds another time, the guest has three options:

- To accept the new wake up time, press **Wake Up**.
- To reject the new wake up time and enter another one, dial the new wake up time and press **Wake Up** to validate the new time.
- To abort the wake up time, press **Rls** or the **Prime DN** key (PDN).

To cancel a wake up request, follow these steps:

1. Press **Wake Up**. The indicator flashes.
2. Dial the octothorpe (#).
3. Press **Wake Up**. The indicator goes off.

To check a wake up request on a telephone with display, follow these steps:

1. Press **Wake Up**. The indicator flashes and the current wake up time appears on the display. If no wake up time is programmed, the display shows a dash (–).
2. Press **Wake Up**. The indicator lights if a wake up time is set.

In each scenario, the Wake Up indicator lights and the display clears, except when the wake up time is aborted and no wake up time was programmed before the abort. In this case, the Wake Up indicator stays off. If a time was programmed before aborting, the previous wake up time is restored, and the indicator is on.

From an attendant console

To program a wake up call from an attendant console, follow these steps:

1. Press A. **Wake Up**. The A. Wake Up, ICI, Ipk, and S indicators light.

   Important:

   If the displayed number is not the number requiring the wake up call, dial the proper number.

2. Press the **octothorpe (#)**. If the A. Wake Up indicator remains on steadily, the dialed number is valid. If it flashes, the number is invalid.
3. Dial the requested wake up time using a 24-hour format. Press A. **Wake Up** again. If the A. Wake Up indicator remains on without flashing, the requested wake up time is acceptable; if it flashes, the time is not acceptable. Enter the new time; if it is acceptable, the indicator goes on without flashing.
4. Press **Rls** to end the procedure.

To cancel a wake up call from an attendant console, follow these steps:

1. Press A. **Wake Up**. The A. Wake Up indicator lights.

If the displayed number is not the number requiring cancellation of the wake up call, dial the proper number.

2. Press the **octothorpe (#)**, then press A. **Wake Up** again. The A. Wake Up indicator goes off and the wake up request is canceled.

   If the indicator flashes quickly, no wake up call was found for the dialed number. Press A. **Wake Up** again.

3. Press **Rls** to end the procedure.

If a guest has not responded after three wake up call attempts, you'll hear a continuous buzz. The indicator will flash quickly. The extension number of the room that has failed to respond will be displayed. Follow these steps:

1. Press A. **Wake Up** to cancel the notification.
2. Press **Rls** to end the procedure.

To Use Multiple Wake Up FFCs

Activate single

The user must dial the Multiple Wake Up Activate (MWUA) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

MWUA H1 H2 M1 M2

Activate repeat (daily)

The user must dial the Multiple Wake Up Repeat Activate (MWRA) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

MWRA H1 H2 M1 M2

Deactivate single

The user must dial the Multiple Wake Up Deactivate (MWUD) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

MWUD H1 H2 M1 M2

Deactivate all

The user must dial the Multiple Wake Up Deactivate (MWUD) FFC:

MWUD H1 H2 M1 M2

Verify

The user must dial the Automatic Wake Up Verify (AWUV) FFC followed by the hour of the wake-up, in 24-hour format, followed by the hour of the next wake-up, in 24-hour format, followed by the minute of the first hour entered followed by the minute of the next hour entered:

AWUV H1 H2 M1 M2

Chapter 69: Automatic Wake Up FFC Delimiter

Contents

This section contains information on the following topics:

Feature description

The Automatic Wake Up Flexible Feature Code Delimiter modifies the user programming interface of the Automatic Wake Up feature, including variations such as Multiple and Repeat Multiple Automatic Wake Up. This modification provides two options for the user: optional delimiter at the end of time entry and optional standard time entry. These options are only applicable to proprietary and analog 2500 telephones.

The optional delimiter at the end of time entry during the activation, deactivation or verification of Automatic Wake Up is an octothorpe (#).

The standard time entry allows a customer to enter standard time when activating Multiple Automatic Wake Up. When activated, a customer can eliminate the leading zero when entering a time. For example, the time seven am can be entered as 700 rather than 0700. The time can still be entered with four digits even if the standard time entry option is selected by the customer.

When activated, this feature provides the user with a response from the system. The response is silence or confirmation by means of a tone or a recorded announcement.

Operating parameters

The feature is applicable to all systems.

If the user enables the delimiter option without enabling the standard time entry option, all four digits (H1H2M1M2) and an octothorpe (#) must be entered for a valid entry.

An octothorpe (#) is the only delimiter accepted to indicate the end of time entry. This delimiter is not programmable.

Feature interactions

Background Terminal

When changes to the wake up timer are initiated by the Background Terminal or user, the wake up time previously entered last is overridden. An octothorpe (#) is not required when entering the Wake up time from a background terminal.

Call Detail Recording

No Call Detail Recording report is generated for Automatic Wake Up calls.

Directory Numbers - Multiple Appearance

For Multiple Appearance Directory Numbers, wake up information is stored, deleted and queried from a DN's first primary appearance terminal number.

Directory Number - Prime Release Key

Pressing the Prime Directory Number or Release key, when programming a Wake up request, cancels the programming sequence. If an invalid timer is entered, the user hears an error tone. If another feature key is pressed during programming, it is ignored by the system.

Room Status

When a guest has either checked in or out, the room status changes. If an AWU request is still active, it is canceled if it is included as part of the Check In/Out option.

Feature packaging

Automatic Wake Up FFC Delimiter requires Flexible Feature Codes (FFC) package 139. The following packages are also required:

- Recorded Announcement (RAN) package 7
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Automatic Wake Up (AWU) package 102

Flexible Tone and Cadences (FTC) package 125 is required if a special error tone rather than overflow is desired for Automatic Wake Up. FTC and Message Intercept (MINT) package 163 is required if a recorded announcement is desired as confirmation from the system after wake up timer has been entered. International Supplementary Features (SUPP) package 131 is required if values other than the default are desired for the inter-digit timer.

Feature implementation

Task summary list

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

1. Table 162: LD 15 - Enable Automatic Wake Up in the Customer Data Block. on page 496
2. Table 164: LD 56 - Set Automatic Wake Up special error tone and configuration tone. on page 497
3. Table 164: LD 56 - Set Automatic Wake Up special error tone and configuration tone. on page 497

Table 162: LD 15 - Enable Automatic Wake Up in the Customer Data Block.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	AWU	Change Automatic Wake Up options.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- AWU	YES	Enable Automatic Wake Up data.
...		
WUD	YES	Wake Up Delimiter. IF WUD = YES then time entry is valid only if user enters octothorpe (#) at end of time digits.
STE	YES	Standard Time Entry prompted only if WUD = YES. This prompt permits three or four digit time entry.

Table 163: LD 57 - Configure Flexible Feature Codes for Automatic Call Wake Up.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FFC	Flexible Feature Codes data block.
CUST	xx	Customer number, as defined in LD 15
FFCT	(NO) YES	Flexible Feature Confirmation Tone.
CODE	AWUA	Auto Wake Up activation code.
- AWUA	xxxx	Auto Wake Up activation code for proprietary and Analog (500/2500 type) telephones. AWUA is prompted until <CR> is entered.
CODE	AWUD	Auto Wake Up deactivation code.
- AWUD	xxxx	Auto Wake Up deactivation code for proprietary and Analog (500/2500 type) telephones. AWUD is prompted until <CR> is entered.
CODE	AWUV	Auto Wake Up verification code
- AWUV	xxxx	Auto Wake Up verification code for proprietary and Analog (500/2500 type) telephones. AWUV is prompted until <CR> is entered.
CODE	MWUA	Multiple Wake Up activation.

Prompt	Response	Description
- MWUA	xxxx	Multiple Wake Up activation code for Analog (500/2500 type) telephones. MWUA is prompted until <CR> is entered.
CODE	MWRA	Repeat Multiple Wake Up activation.
- MWRA	xxxx	Repeat Multiple Wake Up activation code Analog (500/2500 type) telephones. MWRA is prompted until <CR> is entered.
CODE	MWUD	Multiple Wake Up deactivation.
- MWUD	xxxx	Multiple Wake Up deactivation code Analog (500/2500 type) telephones. MWUD is prompted until <CR> is entered.

Table 164: LD 56 - Set Automatic Wake Up special error tone and configuration tone.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	FTC	Flexible Tones and Cadences data block.
TABL	0-31	Flexible Tones and Cadences (FTC) Table Number. To associate a FTC table with trunk route, enter the table number in response to the TTBL prompt in LD 16.
HCCT	YES	Hardware Controlled Cadences and Tone modification of the hardware controlled cadence tone definitions allowed.
- FFCT		Flexible Tone and Cadence.
- - XTON	xxx	Flexible Tone and Cadence confirmation tone. China xxx = 211 North America xxx = 004
- - XCAD	xxx	Cadence code for Firmware Cadence Table (FCAD) as entered at Cadence Number (WCAD) prompt. China xxx = 110 North America xxx = 000
- AWUT		Automatic Wake Up.
- - XTON	xxx	Automatic Wake Up special error tone. China xxx = 214 North America xxx = 007
- - XCAD	xxx	Cadence code for Firmware Cadence Table (FCAD) as entered at Cadence Number (WCAD) prompt. China xxx = 100 North American xxx = 017

Feature operation

The following feature operations occur if the WUD prompt (Wake Up Delimiter) and STE prompt (Standard Time Entry) are set to YES in LD 15. If WUD = YES and STE = NO, then the user must dial all four standard time digits and an octothorpe for a valid entry. If WUD = NO then the STE prompt will not appear. In this case, the prior operation exists and the user is not expected to enter the delimiter (#) at the end of time entry. However, all four time digits must be entered for a valid entry. Table 165: Flexible Feature Codes used in AWU FFC Delimiter feature on page 498 shows the Flexible Feature Codes used in the AWU FFC Delimiter feature.

Table 165: Flexible Feature Codes used in AWU FFC Delimiter feature

Feature	Activation Flexible Feature Code	Deactivation Flexible Feature Code	Verification Flexible Feature Code
Automatic Wake Up (AWU)	AWUA	AWUD	AWUV
Multiple Automatic Wake Up (MAWU)	MWUA	MWUD	AWUV
Repeat Multiple Automatic Wake Up	MWRA	MWUD	AWUV

Flexible Feature Code Automatic Wake Up Activation

To activate Automatic Wake Up from an analog 2500 or a Meridian 1 proprietary telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUA FFC" H1M1M2# or H1H2M1M2#. Get response and go on-hook.

To activate Automatic Wake Up from an analog 500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUA FFC" H1H2M1M2. Get response and go on-hook.

Flexible Feature Code Automatic Wake Up Deactivation

To deactivate Automatic Wake Up from an analog (500/2500) or a Meridian1 proprietary telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUD FFC". Get response and go on-hook.

Flexible Feature Code Multiple Automatic Wake Up Activation

To activate Multiple Automatic Wake Up from an analog 2500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "MWUA FFC" H1M1M2#. Get response and go on-hook.
3. Repeat for up to four wake up times maximum per day.

To activate Multiple Automatic Wake Up time from an Analog 500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "MWAU FFC" H1H2M1M2. Get response and go on-hook.

Flexible Feature Code Multiple Automatic Wake Up Deactivation

To deactivate single wake up time from an analog 2500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "MWUD FFC" H1M1M2# or H1H2M1M2#. Get response and go on-hook.
3. Repeat for other wake up times as necessary.

To deactivate a single wake up time from an analog 500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "MWUD FFC" H1H2M1M2. Get response and go on-hook.
3. Repeat for other wake up times as necessary.

To deactivate all wake up times from an analog 2500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "MWUD FFC" #. Get response and go on-hook.

To deactivate all wake up times from an analog 500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "MWUD FFC" and go on-hook.

Flexible Feature Code Automatic/Multiple Automatic Wake Up Verification

To verify Automatic/Multiple Automatic Wake Up from an analog 2500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUV FFC" H1M1M2# or H1H2M1M2. Get response and go on-hook.
3. Repeat for other wake up times as necessary.

To verify Automatic/Multiple Automatic Wake Up from an analog 500 telephone:

1. Go off-hook. Listen for dial tone.
2. Dial "AWUV FFC" H1H2M1M2. Get response and go on-hook.

Chapter 70: Auxiliary Processor Link

Contents

This section contains information on the following topics:

Feature description

The Auxiliary Processor Link (APL) is a full-duplex asynchronous data link capable of accommodating up to a 4800 baud rate. It is connected to the system through a Serial Data Interface (SDI) port.

This feature is currently used in conjunction with the Integrated Messaging System package and the Automatic Call Distribution (ACD) Dialed Number Identification Service (DNIS) package.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

DNIS Length Flexibility

Expanded DNIS (more than four DNIS digits) is not supported on the APL.

Feature packaging

Auxiliary Processor Link (APL) package 109 has no feature package dependencies.

Feature implementation

To implement this feature, see *Automatic Call Distribution Fundamentals, NN43001-551*.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 71: Auxiliary Signaling

Contents

This section contains information on the following topics:

Feature description

In some situations, customers require special auxiliary devices such as bells, buzzers, or lights to be connected through the system. These devices are activated through a regular 500/2500 Line Card and its associated data block.

Operating parameters

A C4A ringer, or any other special signaling device that can be activated by a 20 Hz ringing signal, can be equipped through the 500/2500 Line Card.

A maximum of five C4A ringers or equivalent devices can be configured on one Terminal Number. This limit depends on the device's impedance to the 20 Hz ringing.

Feature interactions

Mixed DNs

If the DN associated with the signaling device appears on analog (500/2500 type) or Meridian 1 proprietary telephones, the telephone can answer or connect into an active call.

Feature packaging

This feature is included in base system software.

Feature implementation

Table 166: LD 10 - Add new 500 telephone data block.

Prompt	Response	Comment
REQ:	NEW	Add new data block.
TYPE:	500	Analog (500/2500 type) telephone
...		
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
...		

Feature operation

No specific operating procedures are required to use this feature.

Chapter 72: Avaya 1210 IP Deskphone Last Number Redial soft key

Contents

This section contains information about the following topics:

Feature description

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

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

Operating parameters

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

Only the Avaya 1210 IP Deskphone uses the LNR soft key.

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

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

Feature interactions

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

Feature packaging

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

Feature implementation

Use Element Manager to configure LNR on the Avaya 1210 IP Deskphone.

Configuring LNR CLS

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

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

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

Feature operation

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

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

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

Chapter 73: Avaya 3900 Series Digital Deskphones Full Icon Support

This section contains information about the following topics:

Feature description

Using distinct icons and flashing cadences, the Avaya 3900 Series Digital Deskphones Full Icon Support feature informs users of various call states. The icons appear on the LCD next to the Directory Number (DN) keys on the following telephones:

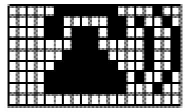
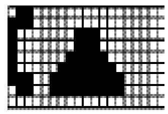
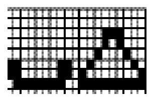
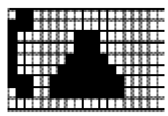
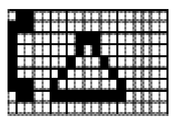
- Avaya 3903 Digital Deskphones (Phase II and Phase III)
- Avaya 3904 Digital Deskphones (Phase II and Phase III)
- Avaya 3905 Digital Deskphones (Phase III)

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

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

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

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

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

Operating parameters

The Avaya 3900 Series Digital Deskphones Full Icon Support feature requires a minimum of Release 9 of the Key-Based Accessory (KBA) module.

Feature interactions

No feature interactions are associated with this feature.

Feature packaging

The Avaya 3900 Series Digital Deskphones Full Icon Support feature requires the following packages:

- Avaya 3900 Series Digital Deskphones Full Icon Support (ICON_PACKAGE) package 397
- Digital Sets (DSET) package 88

Feature implementation

Use LD 17 to enable Avaya 3900 Series Digital Deskphones Full Icon Support.

Table 167: LD 17 — Enable Avaya 3900 Series Digital Deskphones Full Icon Support

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARAM	System parameters.
.....		
ICON	(NO) YES	NO = Disable the Avaya 3900 Series Digital Deskphones Full Icon Support Feature (default). YES = Enable the Avaya 3900 Series Digital Deskphones Full Icon Support Feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 74: Avaya 3900 Series Digital Deskphones Set-to-Set Messaging

This section contains information about the following topics:

Feature description

Use the Set-to-Set Messaging feature to send text messages from one Avaya 3900 Series Digital Deskphones telephone (called party) to another Avaya 3900 Series Digital Deskphones telephone (calling party). On receiving a call, an enabled telephone sends a single text message, as specified from the Applications menu.

The following telephones support Set-to-Set Messaging:

- Avaya 3903 Digital Deskphone
- Avaya 3904 Digital Deskphone
- Avaya 3905 Digital Deskphone

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

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

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

Table 168: Examples of messages text

OUT TO LUNCH
BACK TO WORK: 4 Dec 02
BACK TO OFFICE: Jan 03
WILL REPLY AFTER 1 PM
BACK @ 4:00 PM
NOT IN TODAY
RETURN SOON - 8:10 PM
GONE FOR THE DAY

Operating parameters

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

Using the Applications menu, you can configure up to 10 text messages. However, Avaya 3900 Series Digital Deskphones support only one Set-to-Set message at a time. To activate Set-to-Set Messaging, you must first define at least one message.

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

Feature interactions

Multiple Appearance Redirection Prime/Multiple Appearance Directory Number

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

Feature packaging

Avaya 3900 Series Digital Deskphones Set-to-Set Messaging requires the Set-to-Set Messaging package 380.

Feature implementation

Task summary list:

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

- Table 169: LD 11 on page 515

  Configure Class of Service for Avaya 3900 Series Digital Deskphones Set-to-Set Messaging

- Table 170: LD 15 on page 515

  Modify Set-to-Set Messaging

Table 169: LD 11

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

Table 170: LD 15

Prompt	Response	Description
REQ	NEW CHG	Add new data Change existing datablock information
TYPE:	FTR	Features and options

Prompt	Response	Description
CUST	0-99	Customer number Range for Large System and Avaya Communication Server 1000E (Avaya CS 1000E) system
...		
STS_MSG	(NO) YES	Modify Set-to-Set Messaging
MSG 01	<CR> <text string>	Keeps current message Input the new message to be displayed (up to 24 characters)
...		
MSG 10	<CR> <text string>	Keeps current message Input the new message to be displayed (up to 24 characters)

Feature operation

For more information about the operation of Set-to-Set Messaging, see the *Avaya Digital Deskphones: 3901, 3902, 3903, 3904 User Guide*.

Chapter 75: Avaya 3900 Series Digital Deskphones (Single Site) Virtual Office

Contents

This section contains information on the following topics:

Feature description

For more information on related features and functionality, see “IP Network-wide Virtual Office” and “Incremental Software Management” in *NN43001-106 Features and Services Fundamentals—Book 4 of 6 (I to M)* .

Terms

Host Terminal—A physical Avaya 3903 Digital Deskphone or Avaya 3904 Digital Deskphone at which a user logs in to begin a Virtual Office session.

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

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

Description

The Avaya 3900 Series Digital Deskphones (Single Site) Virtual Office feature allows a user to log in to a designated Avaya 3903 Digital Deskphone or Avaya 3904 Digital Deskphone (Host Terminal) and use an individual telephone configuration (Virtual Set configuration) at that telephone. Calls to the DN assigned to the Virtual Set (the user primary DN) are routed to the Virtual Office Host Terminal where the Virtual Office worker is logged in. The Host Terminal is the physical telephone that a user logs in to as a Virtual Office worker.

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

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

The Avaya 3903 Digital Deskphone and Avaya 3904 Digital Deskphone can be configured as Host Terminals.

A Virtual Office worker is required to log in to a Host Terminal that matches their Virtual Set telephone type. For example, when the individual configuration (the Virtual Set) of a Virtual Office worker is configured as an Avaya 3904 Digital Deskphone, the logon process is blocked, if they attempt to log in to an Avaya 3903 Digital Deskphone Host Terminal.

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

The Avaya 3900 Series Digital Deskphones (Single Site) Virtual Office feature operates on stand-alone Meridian 1 and Avaya Communication Server 1000 (Avaya CS 1000) systems only.

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

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

To log in using the Avaya 3900 Series Digital Deskphones (Single Site) Virtual Office feature, the TN associated with the Host Terminal must be configured with the Virtual Office Login Allowed (VOLA) Class of Service (CLS). The TN associated with the User ID for the logon must be configured with the CLS VOUA (Virtual Office User Allowed). For more information on VOLA and VOUA, see LD 11 and LD 81 in *Avaya Software Input Output - Administration, NN43001-611*.

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

Clearing of the Directory Services Password

With Avaya 3900 Series Digital Deskphones Phase III, the system can clear the Directory Services password when a Virtual Office worker logs in or out of an Avaya 3903 Digital Deskphone or Avaya 3904 Digital Deskphone Host telephone, if Erase List is allowed. The system administrator configures this functionality by defining the Class of Service as Erase List Allowed (ELA) in LD 11 for the Avaya 3903 Digital Deskphone or Avaya 3904 Digital Deskphone Virtual Set.

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

Clearing of the Callers List and Redial List

The contents of the Redial List, Personal Directory, and Call Log are stored on the Avaya 3900 Series Digital Deskphones itself between login sessions.

With Avaya 3900 Series Digital Deskphones Phase III, the system can clear the Redial and Callers lists when a Virtual Office worker logs in or out of an Avaya 3903 Digital Deskphone or Avaya 3904 Digital Deskphone Host telephone. The system administrator configures this functionality by defining the Class of Service as Erase List Allowed (ELA) in LD 11 for the Avaya 3903 Digital Deskphone or Avaya 3904 Digital Deskphone Virtual Set. When the ELA Class of Service is defined, the Callers List and Redial List are automatically cleared when the virtual worker logs in or out. If ELA is not defined, other workers using the same Host telephone can view the Callers and Redial Lists of the previous users.

Automatic Logout for Virtual Office

Avaya 3900 Series Digital Deskphones Phase III introduced automatic logout for Virtual Office workers. If a Virtual Office worker, who is already logged on to telephone A, tries to log on to telephone B, the system automatically logs the Virtual Office worker off at telephone A and logs them on to telephone B (provided that the Virtual Office worker enters the correct login password). The system administrator enables this functionality in LD 15 at the Virtual Office Automatic Logout (VO_ALO) prompt.

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

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

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

Speed Call for Virtual Office

With Avaya 3900 Series Digital Deskphones Phase III, Avaya 3900 Series Digital Deskphones support Speed Call (SCU/SCC) and System Speed Call (SSU only) on Virtual Set TNs.

Operating parameters

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

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

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

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

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

Feature interactions

Speed Call

With Avaya 3900 Series Digital Deskphones Phase III, Avaya 3900 Series Digital Deskphones support Speed Call (SCU/SCC) and System Speed Call (SSU only) on Virtual Set TNs.

Feature packaging

Avaya 3900 Series Digital Deskphones (Single Site) Virtual Office requires the following packages:

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

Feature implementation

Task summary list

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

1. Table 171: LD 97 on page 521

   Configure a Phantom loop for Virtual Set.
2. Table 172: LD 57 on page 522

   Configure Virtual Office Flexible Feature Codes.
3. Table 173: LD 11 on page 522

   Configure the Virtual Set with Erase List Allowed.
4. Table 174: LD 81 on page 522

   Print a list or count of Virtual Office terminals.
5. Table 175: LD 20 on page 523

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

Table 171: LD 97

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

Table 172: LD 57

Prompt	Response	Description
REQ:	NEW	Add new data block information.
	CHG	Change data block information.
	OUT	Remove data block information.
	END	Exit data block overlay program.
TYPE	FFC	Flexible Feature Codes data block.
...	...	
CODE	VTLN	FFC for Virtual Set login.
	ALL	Every FFC is prompted.
	<CR>	No further prompt; returns to REQ.
VTLN	xxxx	Virtual Set login code.
	<CR>	Returns to "CODE"
CODE	VTLF	FFC type for Virtual Set logout.
	ALL	Every FFC is prompted.
	<CR>	No further prompt; returns to REQ.
VTLF	xxxx	Virtual Set logout code.
	<CR>	Returns to "CODE" A Phantom TN cannot be moved or copied.

Table 173: 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, Avaya CS 1000 Media Gateway 1000B (Avaya MG 1000B), and Avaya Communication Server 1000E (Avaya CS 1000E) system, where l = loop, s = shelf, c = card, u = unit.
CLS	(ELD) ELA	Erase Lists (Denied) Allowed.

Table 174: LD 81

Prompt	Response	Description
REQ	LST	Print a list of telephones.

Prompt	Response	Description
	CNT	Print a count of telephones.
CUST	xx	Customer number, as defined in LD 15.
...	...	...
FEAT	aaa	Designate a feature mnemonic.
	3900	Print Avaya 3900 Series Digital Deskphones-type telephones, including Virtual Sets and Host Terminals.
	DCFW	Print default call forward for Phantom TNs, including Virtual Sets.
...	...	...

Table 175: LD 20

Prompt	Response	Description
REQ:	PRT	Print data block for the requested terminal type(s).
	LTN	List Terminal Numbers of the requested terminal type(s).
TYPE:	a...a TNB	Telephone type, where a...a = 3903V (Avaya 3903 Digital Deskphone Virtual Set), 3904V (Avaya 3904 Digital Deskphone Virtual Set), 3903H (Avaya 3903 Digital Deskphone Host Terminal), or 3904H (Avaya 3904 Digital Deskphone Host Terminal). The only telephone types of the Avaya 3900 Series Digital Deskphones that can be configured as a Virtual Set or Host Terminal are the Avaya 3903 Digital Deskphone and Avaya 3904 Digital Deskphone. The Print TNB and List TNB requests always show the logged-off TNB data. In logged-in state, an indication of the logged-in TN ("HOST TN" or "VIRTUAL TN") is added.

Feature operation

For more information on the operation of this feature, see the *Avaya Digital Deskphones: 3901, 3902, 3903, 3904 User Guide*.

Chapter 76: Avaya Integrated DECT

Contents

This section contains information on the following topics:

Feature description

Avaya Integrated DECT (DECT) is an application on the system that allows digital wireless capabilities. With DECT, users can travel around their work sites while answering a call, making a call, continuing a call, or transferring a call. For detailed information, see Avaya DECT Fundamentals.

Operating parameters

DECT includes a DECT Mobility Card (DMC) and a DECT Mobility Card Expander (DMC-E). These cards exist in an Intelligent Peripheral Equipment Module of the system. The cards provide and manage the radio network used in wireless service.

On an Avaya Communication Server 1000M (Avaya CS 1000M) or Meridian 1 PBX 11C Cabinetsystem, DECT supports a maximum of 630 users. An Avaya CS 1000M Chassisor Meridian 1 PBX 11C Chassissupports a maximum of 96 DECT users. Large Systems support a maximum of 1024 users.

Feature interactions

DECT does not require DTI programming in LD 73.

Feature packaging

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

Feature implementation

Task summary list

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

1. Table 176: LD 10 on page 526

   Configure a DECT telephone
2. Table 177: LD 73 on page 527

   Configure DECT pad values

Table 176: LD 10

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

Prompt	Response	Description
WRLS	(NO) YES	Indicates that this TN corresponds to a portable personal telephone or DECT Handset 4060. Only offered if the MCMO package is equipped.
WYTP	(MCMO) DECT	Wireless type assigns the TN to DECT cards. The WYTP prompt appears when WRLS = YES.
CLS	(CNDD) CNDA	Allows the user to see calling or called name associated with the number dialed if CPND is set up for the customer associated with the portable personal telephone. Permitted only if WRLS = YES.
	(MCRD) MCRA	Multiple Call Arrangement (denied) allowed. Allows privacy on analog (500/2500-type) telephones including both portable and wireline telephones. Only offered if the MCMO package or SUPP package is equipped.
	(DTN)	Default digit signaling used by portable personal telephone.

Table 177: LD 73

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data
TYPE	PRI2 PRI	2.0 Mbps/1.5 Mbps PRI data block
FEAT	PAD	Pad category
PDCA	1-16	Pad category table
...		
BRIT	Rx Tx	BRI trunk.
MCM	Rx Tx	DECT pad value, where: R = Receive T = Transmit, and x = pad value (0-26).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 77: B34 Codec Static Loss Plan Downloading

Contents

This section contains information on the following topics:

Feature description

This feature provides software support for Static Loss Plan Downloading to the B34 codec. A codec is a device on an Intelligent Peripheral Equipment (IPE) card which encodes incoming transmission data from analog to digital, and decodes outgoing transmission data from digital to analog.

The B34 codec is a four-channel codec providing 32 programmable loss values in 0.5 dB steps in both the transmit and receive directions. The B34 allows transmission parameters, which can be downloaded to the IPE unit, to be changed by software. Since the loss and level requirements differ from country to country, this allows system compliance to the different transmission plans used in the world markets using a single codec.

The selected coded levels are downloaded to each unit based on the unit's port type classification at initialization, configuration, or enable time. This is referred to as static downloading. These levels will be used for all call connections involving that unit. The B34 Codec Static Loss Plan Downloading feature is used on systems where a single loss setting is sufficient for all types of call connections.

Some markets, however, require adjustments on the loss setting depending on the call connection. This is referred to as Dynamic Pad Switching, or Dynamic Loss Switching, and is addressed by the Dynamic Loss Switching feature, which is described elsewhere in this publication. The B34 Codec Static Loss Plan Downloading feature provides the basis for the Dynamic Loss Switching feature; if Dynamic Loss Switching is enabled for a system, Static Loss Plan Downloading is suppressed on that same system.

The transmission plan for each country follows the European Telecommunications Standards Institute (ETSI) standard of loss values (referred to as "new values"), or existing values (loss values currently provided by existing cards in ETSI countries). New IPE cards must be capable of accepting these existing values for use in existing systems, so as to maintain port-to-port loss integrity.

Typically, existing (pre-Phase 8B) systems do not require flexible B34 equipped IPE cards unless their loss plans change; these systems use the existing loss plans. New systems installed with Phase 8B software contain only flexible B34 equipped IPE cards, and can use either existing loss plans or the ETSI loss plans. Systems equipped with both flexible B34 equipped IPE cards and non-B34 equipped IPE cards require type approval to be secured under existing loss plan values.

The Static Loss Plan Download feature allows the selection of a loss plan table which is either compliant with the old or the new loss plan for various countries. The feature is supported on international IPE analog trunk cards (XCOT, XFCOT, XDID, XCO/XDID, XFEM, or any trunk configured with XTRK type of XCOT, XDID, or XFEM) with the right firmware support. In special situations and with the right authorization, a customized table may be defined.

When selecting a loss plan table, it is important to verify whether the existing or ESTI mode is to be exhibited by the system (the "Feature implementation" section explains how to install a loss plan table using LD 97). A Service Change interface allows an existing or ETSI table to be selected by specifying a loss plan table number. If the loss plan needs to be upgraded in the field or if a newly defined loss plan has to be installed, a service change may be performed by an authorized craftsperson to enter a table of customized loss plan values for each port type, or to customize a pre-defined table by changing the table values. The table can then be downloaded upon any of the following conditions:

- at system initialization for all units
- when a trunk or line card, or trunk or line unit is enabled
- when the XPEC is enabled
- when the IPE shelf is enabled
- when a configured card is reset
- after a trunk unit has undergone a "NEW", "CHANGE", or "MOVE" operation using LD 14 or LD 10

There is no mechanism to indicate whether or not an IPE card is equipped with the B34 codec. Loss plan messages are downloaded to all IPE cards in hybrid systems, whether or not they are equipped with the B34 codec. Typically, there are three vintages of firmware used in the field:

- non-B34 codec equipped cards
- hardcoded B34 equipped cards
- flexible B34 equipped cards

There are two versions of the flexible B34 equipped cards, a flexible 7C software compatible B34 equipped card and a flexible 8B software compatible B34 equipped card. The hardcoded and 7C software compatible versions of the B34 equipped cards have country-unique loss value defaults. The flexible 8B software compatible B34 equipped card have universal B34 default loss values, which do not meet any country-specific requirements.

The flexible 7C software compatible B34 equipped card and the flexible 8B software compatible B34 equipped card both recognize the new B34 (type 12) messages, as well as the old static pad switching (type 5) messages; the type 12 messages take precedence. The hardcoded B34 cards only recognize the type 5 messages.

The flexible 7C B34 equipped cards are forward compatible with the new software; the hardcoded B34 cards are not. The flexible 8B B34 equipped cards are not backwards compatible to systems running older versions of software.

Operating parameters

A system must be configured with one or more IPE cards equipped with a B34 codec and firmware supporting software downloading. It is the responsibility of the installer to verify that the IPE cards used are compliant with the download messages used by this feature.

XFALC (flexible analog line card) is compatible with the download messages supporting Static Loss Plan Download.

Since the flexible 8B B34 equipped cards are not backwards compatible to systems running older versions of software, the following upgrade strategy should be followed:

- Systems running software Phase 7C or earlier, and upgrading to Phase 8B software, do not require the new flexible B34 IPE cards if the transmission plan remains the same. These systems may be equipped with a mix of hardcoded B34 IPE cards and new flexible B34 IPE cards; if changing to the new ETSI loss plan, all hardcoded B34 IPE cards must be retrofitted with the new flexible B34 IPE cards.
- Systems changing to a new ETSI loss plan must use the new flexible B34 IPE cards as well as Phase 8B static parameter download software; a hardware retrofit and a software upgrade are also necessary.
- Newly installed systems will use the new flexible B34 IPE cards.

New flexible B34 equipped XFALC (flexible analog line cards) support Static Loss Plan Downloading using B34 messages. New flexible B34 equipped XFALCs installed in a Phase

7C software environment do not receive download messages, but use the firmware-defined default.

A distinction must be made between long and short lines on ALC units, and to download loss plan values based on this setting.

Feature interactions

Alternative Loss Plan

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

B34 Dynamic Loss Switching

B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching. Both features share the same definition of port types and use the same base-level table.

When B34 Dynamic Loss Switching is enabled, the Static Download messages to the analog trunk cards are suppressed. Static download to analog line cards continues.

B34 codec static loss download. Since the B34 Dynamic Loss Switching is dependent on B34 Codec Static Loss Download, B34 Codec Static Loss Download must be enabled when B34 Dynamic Loss Switching is enabled. The port types defined for B34 Dynamic Loss Switching are a subset of the port types defined for B34 Codec Static Loss Download.

Also, the base level table used by B34 Codec Static Loss Download is also used by B34 Dynamic Loss Switching. Since B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching, B34 Codec Static Loss Download is enabled when B34 Dynamic Loss Switching is enabled. When B34 Dynamic Loss Switching is enabled, the following operations concerning trunk cards are suppressed:

- During initialization, B34 Codec Static Loss Downloading to trunk cards is suppressed, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- In LDs 32 and 36, B34 Codec Static Loss Download is suppressed on enabling the trunk card or unit, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- When resetting the cards, B34 Codec Static Loss Download is suppressed. Downloading continues to analog line cards.

- In LD 14, B34 Codec Static Loss Download is suppressed.
- In LD 10, B34 Codec Static Loss Download is suppressed.

When B34 Dynamic Loss Switching is disabled, all B34 Codec Static Loss Downloads to trunks are suppressed. This introduces the danger of having some cards in the system which are not set with the proper loss levels, since the system has been changed from a dynamic mode to a static mode without activating the download of the static messages. To highlight this change, a SCH5842 error message is generated, indicating to the craftsperson that B34 Dynamic Loss Switching is disabled, and that B34 Static Loss Downloading is now in effect and that a download should be activated by system initialization or SYSLOAD.

When B34 Dynamic Loss Switching is enabled, all B34 Codec Static Loss Download audit messages to trunk cards are suppressed.

Conference

When a conference connection is established, no pads are switched in on the trunk side; any extra loss that is required is provided by the conference circuit based on an algorithm which takes into account the number of lines and trunks.

Digital Trunk Interface (DTI) Pad Switching

Pad switching for DTI applications is done dynamically, based on the far end's port type. On the DTI side, a loss value is switched on the receive and transmit side, depending on the far end's port type. If the far end is analog, a pad is switched in or out; if the far end is digital, a zero loss is switched in, so that the relative loss is taken care of only on one side. Connection between DTI/PRI and XDID, XFCOT, and XFEM trunks is not supported, since DTI pad switching does not take care of these trunk types.

DTI2 Pad Switching

Pad switching for DTI2 applications is done dynamically, based on the far end's port type. On the DTI side, a loss value is switched on the receive and transmit side, depending on the far end's port type. The far end side is handled by the normal operation for the trunk type. That is, if the far end is DTI, it is handled according to the DTI pad switching. If it is otherwise, it is handled by the configured matrices. No messages are sent for XDID, XFCOT, and XFEM trunk types. For XUT and XEM trunk types, the loss equivalent to pad out is switched in. For XDID, XFCOT, and XFEM trunks, the base level (static) value is switched in when connected to the DTI2 trunk types.

GEC Plessy hardware

No losses are sent to XCOT, XDID, and XFEM trunk cards when these cards are connected to GEC hardware, since there is no dynamic switching done for them. On the GEC hardware side of such a connection, the pads are switched in according to the type of trunk (near end) as opposed to what it is connected (far end); therefore, the loss is switched in regardless of whether the connection is to XCOT, XDID, and XFEM trunk cards or other types of cards.

Intelligent Peripheral Equipment Completion

Whenever a TIE/LDR trunk is configured on an XIDID card, for Static Loss Plan Download (SLPD)/Dynamic Loss Switching (DLS), loss/level is downloaded/switched to an XDID card with the type 12 message. Depending on the Class of Service configured, Non-Transmission Compensated (NTC), Transmission Compensated (TRC), or Via Net Loss (VNL), the TIE unit will be mapped to the following B34 port types: B34 T2WN, B34 T2WT, or B34 T2WV.

ISDN Basic Rate Interface

It is possible to switch in loss on the ISDN BRI side, based on port types.

MFE/MFC Pads

The Alternative Loss Plan feature allows trunks to be configured so as to have pads switched in when an MFS sender/receiver is equipped. For such a configuration, the following occurs for B34 port types:

- Pads are switched in for outgoing calls (the trunk is the originator).
- Pads are switched in, if in the dialing state, for incoming calls (the trunk is the terminator).

Feature packaging

B34 Codec Static Loss Plan Downloading requires Intelligent Peripheral Equipment (XPE) package 203.

Feature implementation

The loss level tables are configured in LD 97. The craftsperson must have an authorized password to configure the loss tables, but printing of the tables can be performed without the password.

Table 178: LD 97 - Configure a loss plan table.

Prompt	Response	Description
TYPE	LOSP	The type branch for the system loss plan table.
STYP	(PRED) CSTM DISL	The type of B34 static loss plan table to be used to download B34 programmable loss codes. Enter PRED if a numbered pre-defined static loss plan is to be used. Enter CSTM to customize an existing static loss plan table by modifying one or more existing entries, or to create a new table by entering new values to all entries. Enter DISL to disable static loss plan downloading.
PWD2	xxxx	Enter the level 2 administrator password. Note that this is prompted only when STYP=DISL or STYP=CSTM. If STYP=DISL, and the proper password is entered, then the next prompt is REQ. If STYP=CSTM, and the proper password is entered, then the next prompts are the PORTTYPES (for example, COTS, COTL). If the password entered is incorrect, an existing error message, SCH523, SCH525, SCH526 will be issued and PWD2 will be re-prompted.
TNUM	1-25	Prompted only if PRED was entered in response to the STYP prompt above. Enter the number for the required pre-defined static loss plan.
COTS	Rx Tx	Prompted only if the response to the STYP prompt above was CSTM. COT short line. Enter the coded input/output relative levels in the receive (Rx) direction and in the transmit (Tx) direction, for this port type. The input range of Rx and Tx for port types associated with trunks is 8-39 and 0-31 respectively; the input range of Rx and Tx for port types associated with analog lines is 0-31 and 8-39 respectively.
COTL	Rx Tx	COT long line. The same definition applies as for COTS.
DIDS	Rx Tx	DID/DOD short line. The same definition applies as for COTS.

Prompt	Response	Description
DIDL	Rx Tx	DID/DOD long line. The same definition applies as for COTS.
T2WT	Rx Tx	TIE, 2 wire, Class of Service TRC. The same definition applies as for COTS.
T2WN	Rx Tx	TIE, 2 wire, Class of Service NTC. The same definition applies as for COTS.
T2WV	Rx Tx	TIE, 2 wire, Class of Service VNL. The same definition applies as for COTS.
T4WT	Rx Tx	TIE, 4 wire, Class of Service TRC. The same definition applies as for COTS.
T4WN	Rx Tx	TIE, 4 wire, Class of Service TRC. The same definition applies as for COTS.
T4WV	Rx Tx	TIE, 4 wire, Class of Service VNL. The same definition applies as for COTS.
PAGT	Tx	TIE, E&M 2 paging trunk. The same definition for Tx applies as for COTS. Note that there is no loss value associated with this trunk type in the receive (Rx) direction.
RANR	Rx	Recorded Announcement Route. The same definition for Rx applies as for COTS. Note that there is no loss value associated with this trunk type in the transmit (Tx) direction. Enter the Coded Receive (A/D) Input/ Output level, where Rx = 8-39.
ALUS	Rx Tx	ALC unit short line (SHL) Class of Service. Enter the coded input/output relative levels in the receive (Rx) direction and in the transmit (Tx) direction, for this port type. The input range of Rx and Tx for port types associated with analog lines is 0-31 and 8-39 respectively.
ALUL	Rx Tx	ALC unit long line (LOL) Class of Service. Enter the coded input/output relative levels in the receive (Rx) direction and in the transmit (Tx) direction, for this port type. The input range of Rx and Tx for port types associated with analog lines is 0-31 and 8-39 respectively.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 78: B34 Dynamic Loss Switching

Contents

This section contains information on the following topics:

Feature description

A codec is a device on an Intelligent Peripheral Equipment (IPE) card which encodes incoming transmission data from analog to digital, and decodes outgoing transmission data from digital to analog. The B34 codec is a four-channel codec providing 32 programmable loss values in 0.5 dB steps in both the transmit and receive directions. The B34 allows transmission parameters, which have been downloaded to the IPE unit, to be changed by software. Since the loss and level requirements differ from country to country, this allows system compliance to the different transmission plans used in the world markets using a single codec. The selected coded levels are downloaded to each unit based on the unit's port type classification. This is referred to as static downloading. These levels will be used for all call connections involving that unit. The B34 Codec Static Loss Plan Downloading feature is, therefore, used on systems where a single loss setting is sufficient for all types of call connections (this feature is described elsewhere in this publication).

The Dynamic Loss Switching feature provides loss switching on international IPE analog trunks cards (XCOT, XFCOT, XDID, XCO/XDID, XFEM, or any trunk configured with XTRK type of XCOT, XDID, or XFEM).

Typically, there are different vintages of firmware in the field:

- hard-coded B34 firmware, which is hardcoded with country-specific defaults, ignores B34 type 12 messages, and accepts (where applicable) Short Line/Long Line configuration type 5 messages
- flexible B34 firmware with country-specific defaults, which is firmware that is coded with country-specific defaults, accepts (where applicable) Short Line/Long Line configuration type 5 messages, and accepts B34 type 12 messages which override any accepted Short Line/Long Line configuration type 5 messages
- flexible B34 firmware with universal defaults, which is firmware that is coded with a universal B34 loss value default, may or may not ignore Short Line/Long Line configuration type 5 messages, and accepts B34 type 12 messages which override any accepted Short Line/Long Line configuration type 5 messages

To obtain the full functionality of B34 Dynamic Loss Switching, only the two flexible vintages of firmware can be used.

Every time a new connection is established, the following process is followed to determine if and how to adjust the loss involved in the connection:

- the port type of the originator and terminator is determined, based on the configurations of the originator and terminator, respectively
- this port type is used as a row index (originator) and column index (terminator) into a connection matrix, to determine the following:
    - whether to switch the pad in or out for the originator receive direction whether to switch the pad in or out for the originator transmit direction
    - whether to switch the pad in or out for the terminator receive direction
    - whether to switch the pad in or out for the terminator transmit direction
- a message conveying this information is then sent to the originator and terminator, if they are affected port types.

The B34 Dynamic Loss Switching feature, configured on a system basis, introduces flexibility in the loss values to be switched. Where previously the loss values were hardcoded on the analog trunk cards, they are now software-configurable on a per-system basis. The loss switching is still controlled by a connection matrix defined for specific markets. This matrix cannot be changed. The loss levels to be used are configured in a base-level table and alternative-level table in LD 97. The base level table is the same as the one implemented and used by the B34 Static Loss Plan Downloading feature; the alternative level table is a parallel table configured for the B34 Dynamic Loss Switching feature.

These new port types reside on the international IPE cards with flexible B34 firmware and the B34 codec. They have to be distinguished from existing port types because of the different manner in which they are informed of the base level/alternative level information.

Operating parameters

A system must be configured with one or more IPE card equipped with a B34 codec and firmware supporting software downloading. It is the responsibility of the installer to verify that the IPE cards used are compliant with the download messages used by this feature.

The B34 Codec Static Loss Plan Downloading feature must be equipped, since the B34 Dynamic Loss Switching feature uses its base level table.

Since the flexible 8B B34 equipped cards are not backwards compatible to systems running older versions of software, the following upgrade strategy should be followed:

- Systems running software Phase 7C or earlier, and upgrading to Phase 8B software, do not require the new flexible B34 IPE cards if the transmission plan remains the same. These systems may be equipped with a mix of hardcoded B34 IPE cards and new flexible B34 IPE cards; if changing to the new European Telecommunications Standards Institute (ETSI) loss plan, all hardcoded B34 IPE cards must be retrofitted with the new flexible B34 IPE cards.
- Systems changing to a new ETSI loss plan must use the new flexible B34 IPE cards as well as Phase 8B static parameter download software; a hardware retrofit and a software upgrade are also necessary.
- Newly installed systems will use the new flexible B34 IPE cards.

XFALC (Flexible Analog Line Card) is compatible with the download messages supporting Static Loss Plan Downloading. XFALC is not supported in Dynamic Loss Switching.

Connection matrixes are supported for Australia, New Zealand, and Italy. No other countries are supported with this feature.

New flexible B34 equipped XFALC (flexible analog line cards) support Static Loss Plan Downloading using B34 messages. New flexible B34 equipped XFALCs installed in a Phase 7C software environment do not receive download messages, but use the firmware-defined default.

A distinction must be made between long and short lines on Analog Line Cards (ALC), and to download loss plan values based on this setting.

Feature interactions

Alternative Loss Plan

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

B34 Codec Static Loss Download

B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching. Both features share the same definition of port types and use the same base-level table.

When B34 Dynamic Loss Switching is enabled, the Static Download messages to the analog trunk cards are suppressed. Static download to analog line cards continues.

B34 codec static loss download. Since the B34 Dynamic Loss Switching is dependent on B34 Codec Static Loss Download, B34 Codec Static Loss Download must be enabled when B34 Dynamic Loss Switching is enabled. The port types defined for B34 Dynamic Loss Switching are a subset of the port types defined for B34 Codec Static Loss Download.

Also, the base level table used by B34 Codec Static Loss Download is also used by B34 Dynamic Loss Switching. Since B34 Codec Static Loss Download is a prerequisite for B34 Dynamic Loss Switching, B34 Codec Static Loss Download is enabled when B34 Dynamic Loss Switching is enabled. When B34 Dynamic Loss Switching is enabled, the following operations concerning trunk cards are suppressed:

- During initialization, B34 Codec Static Loss Downloading to trunk cards is suppressed, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- In LDs 32 and 36, B34 Codec Static Loss Download is suppressed on enabling the trunk card or unit, so that loss levels do not change in case there are active calls. Downloading continues to analog line cards.
- When reseating the cards, B34 Codec Static Loss Download is suppressed. Downloading continues to analog line cards.
- In LD 14, B34 Codec Static Loss Download is suppressed.
- In LD 10, B34 Codec Static Loss Download is suppressed.

When B34 Dynamic Loss Switching is disabled, all B34 Codec Static Loss Downloads to trunks are suppressed. This introduces the danger of having some cards in the system which are not set with the proper loss levels, since the system has been changed from a dynamic mode to a static mode without activating the download of the static messages. To highlight this change, a SCH5842 error message is generated, indicating to the craftsperson that B34 Dynamic Loss

Switching is disabled, and that B34 Static Loss Downloading is now in effect and that a download should be activated by system initialization or SYSLOAD.

When B34 Dynamic Loss Switching is enabled, all B34 Codec Static Loss Download audit messages to trunk cards are suppressed.

Conference

When a conference connection is established, no pads are switched in on the trunk side; any extra loss that is required is provided by the conference circuit, based on an algorithm which takes into account the number of lines and trunks.

Digital telephone transmission parameters

The following static parameters, which do not change on a connection basis, can be changed using LD 17:

- sidetone objective loudness rating
- receive objective loudness rating
- transmit objective loudness rating
- handsfree receive objective loudness rating
- handsfree transmit objective loudness rating
- handsfree receive objective loudness rating

Digital Trunk Interface (DTI) Pad Switching

Pad switching for DTI applications is done dynamically, based on the far end's port type. On the DTI side, a loss value is switched on the receive and transmit side, depending on the far end's port type. If the far end is analog, a pad is switched in or out; if the far end is digital, a zero loss is switched in, so that the loss is taken care of only on one side. Connection between DTI/PRI and XDID, XFCOT, and XFEM trunks is not supported, since DTI pad switching does not take care of these trunk types.

The far end side is handled by the normal operation for the trunk type: that is, if the far end is DTI, it is handled according to the DTI pad switching, if it is otherwise, it is handled by the configured matrices. No messages are sent for XDID, XFCOT, and XFEM trunk types. For XUT and XEM trunk types, the loss equivalent to pad out is switched in. For XDID, XFCOT, and XFEM trunks, the base level (static) value is switched in when connected to the DTI2 trunk types.

Echo Suppression

When the echo suppresser is turned on for XEM and XFEM trunks, the pad is switched to out. For XEM and XFEM trunks with B34 port types, the base loss level for the affected port type is switched in to match the operation of switching out the pad.

GEC Plessy Hardware

No losses are sent to XCOT, XDID, and XFEM trunk cards when these cards are connected to GEC hardware, since there is no dynamic switching done for them. On the GEC hardware side of such a connection, the pads are switched in according to the type of trunk (near end) as opposed to what it is connected (far end); therefore, the loss is switched in regardless of whether the connection is to XCOT, XDID, and XFEM trunk cards or other types of cards.

ISDN Basic Rate Interface

It is possible to switch in loss on the ISDN BRI side, based on port types.

MFE/MFC Pads

The Alternative Loss Plan feature allows trunks to be configured so as have pads switched in when am MFC sender/receiver is equipped. For such a configuration, the following occurs for B34 port types:

- pads are switched in for outgoing calls (the trunk is the originator), or
- pads are switched in, if in the dialing state, for incoming calls (the trunk is the terminator).

Off Premise Extension Pad Switching

Pads can be switched on an Off Premise Extension card depending on the type of connection.

XCOT, XFEM, and XDID Cards

XCOT, XFEM, and XDID cards are the suite of international IPE cards which are configured under the XTRK prompt in LD 14. The cards in this suite include XDID/DOD, XFCOT, XFEM, XDID, and XCOT. When B34 Dynamic Loss Switching is enabled, these cards receive B34 messages. Since certain markets do not desire this functionality, B34 Dynamic Loss Switching should not be enabled.

During lamp audit for active calls on XCOT, XFEM, and XDID cards, a type 5 message for pad switching is sent to these cards, based on their configuration. When B34 Dynamic Loss Switching is enabled, the type 5 message is not sent; instead, a B34 message is sent, based on the last loss switching message sent for that call.

XEM and XUT Cards

XEM and XUT cards are the suite of North American IPE cards which are configured under the XTRK prompt in LD 14. The cards in this suite include XUTJ, XUT Hong Kong, XEM, and XUT. When B34 Dynamic Loss Switching is not enabled, there is no change in the operation of pad switching on these cards. When B34 Dynamic Loss Switching is enabled, the expanded portion of the connection matrix is used to determine the processing on the XEM/XUT side of the call and on the B34 port type side of the call. When a decision is made, it is communicated using a B34 message.

Feature packaging

B34 Dynamic Loss Switching requires the following packages:

- International Supplementary Features (SUPP) package 131
- Limited Access to Overlays (LAPW) package 164
- Intelligent Peripheral Equipment (XPE) package 203

Feature implementation

The base and alternate tables are configured in LD 97. The connection matrix is selected in LD 15.The craftsperson must have an authorized password to configure the loss tables. Printing of the tables can be performed without the password.

The system must be configured with the Limited Access to Overlays (LAPW) package, and the craftsperson must have an authorized password.

Table 179: LD 97 - Configure a loss plan table.

Prompt	Response	Description
REQ	CHG	Change loss plan table.

Prompt	Response	Description
TYPE ...	LOSP	The type branch for the system loss plan table. Enter LOSP.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 79: Background Terminal

Hospitality and health care personnel use Background Terminal (BGD) to enter, retrieve, and modify data associated with the following features:

- Automatic Wake Up (AWU)
- Room Status (RMS)
- Message Registration (MR)
- Call Party Name Display (CPND)

BGD helps monitor system operations by providing a visual display of information changes, hard-copy backup, and traffic statistics.

For more information, see *Hospitality Features Fundamentals* (NN43001-553).

Chapter 80: Bandwidth Management Support for Network Wide Virtual Office

Contents

This section contains information on the following topics:

Feature description

The Bandwidth Management Support for Network Wide Virtual Office (NWVO) feature allows the assignment of the same Virtual Private Network Identifier, VPNI, to all Call Servers in a network, so that the entire network can be identified by one VPNI number. It allows the assignment of the same Bandwidth Zone number to different Call Servers and to have an INTRAZONE policy between them. At the same time, this feature does not interfere with the existing functionality of the Bandwidth Management (BWM) feature, because previous bandwidth configuration rules are still supported.

This feature extends the meaning of the Virtual Private Network Identifier (VPNI). In previous releases, VPNI was used to identify one customer system (Main Office (MO) switch + Branch Office (BO) switches connected to this MO switch). In CS 1000 Release 5.0 and later, a VPNI number identifies the entire customer network that includes all MO and BO switches.

This feature introduces a new Current Zone field for the IP Deskphones to distinguish between the Bandwidth Zone number configured for the IP Deskphone and the current, real, zone number that is not configurable, but is changed dynamically by Virtual Office feature operation. The Current Zone field is used in the bandwidth calculation routines, rather than the Configured

Zone field, to have correct bandwidth calculation in case of NWVO call scenarios. The customer can check the value of the Current Zone for IP Deskphones using the LD 20 PRT command; as this value is not configurable, no changes are made to LD 11.

Operating parameters

The following are operating parameters for this feature:

- Only IP Phones with the "IP Client cookies" feature enabled are supported
- The correct BWM calculation for a Virtual Office (VO) IP Phone is possible only within the home Customer network (when all the systems are configured with the same VPNI number).
- The maximum number of Bandwidth Zones that can be configured on the Call Servers within the home customer network is limited to 255 (1-255).
- The interaction between Communication Server 1000 (CS 1000) Release 5.0 and earlier CS 1000 releases is not supported. Therefore, the correct bandwidth calculation is not provided if a remote VO logon is performed from/to CS 1000 with a software release older than Release 5.0. In this case, the old bandwidth calculation method is used.
- If two or more Call Servers have the same VPNI and Bandwidth Zone configured, the usage in this zone cannot be synchronized between these Call Servers.
- The feature supports only one customer. If more than one customer is configured, the customer with the lowest customer number is supported.

Assumptions

The following configuration is used:

- Bandwidth zones are allocated on a Network-wide basis.
- All zones in the network are configured on all the Call Servers.
- All Call Servers within the network will have the identical copy of the zone table and each individual zone policy.
- Call Servers that belong to different networks will have different VPNI numbers. Each network has its individual VPNI numbers.
- The VPNI number will be configured as the same for all Call Servers within the network.
- The identical copy of the zone table within the network should be configured manually unless a synchronization mechanism is introduced.

Feature interactions

The following feature interactions exist:

Interaction with Zone-based Digit Manipulation

A new prompt is added to the Customer Data Block (CDB) to configure the zone (Current or Configured) to be used for Zone-based Digit Manipulation feature. It allows VO users to use either local PSTN connections of the Call Server where they are physically located (Current Zone), or use remote PSTN connections of the Call Server where their VO TNs are configured (Configured Zone), depending on customers' preferences.

Interaction with Time and Date

A new prompt is added to the CDB to configure the zone (Current or Configured) to be used for Time and Date feature. It allows the VO user to have local time and date on the display of the IP set used for VO logon to home CS.

Interaction with Off-Hook Alarm Security

Off-Hook Alarm Security feature uses information about Current Zone field rather than the Configured Zone field for their operation.

Feature packaging

This feature needs two packages to be equipped on the system:

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

Feature implementation

Task summary list

The following is a list of all tasks in this section:

Table 180: LD 15 – Define VPNI number in Customer Data Block

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	NET_DATA	Networking.
-VPNI	1-16283	Virtual private network identifier.

Table 181: LD 117 – Define Zone Data

Command	Description
NEW ZONE <zoneNumber> <intraZoneBandwidth> <intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy> [<zoneIntent> <zoneResourceType>]	Define a new Zone with parameters. All parameters must be entered: - zoneNumber from 0 to 255. - intraZoneBandwidth from 0 to 0.1Mbps. - intraZoneStrategy is the intrazone preferred strategy where BQ is Best Quality or BB is Best Bandwidth. - interZoneBandwidth from 0-0.1Mbps. - interZoneStrategy is the interzone preferred strategy where BQ is Best Quality and BB is Best Bandwidth. - zoneIntent is the type of zone, where MO is Main Office zone, BMG is Branch Media Gateway zone and VTRK is Virtual Trunk zone. - zoneResourceType is resource Intrazone preferred strategy, where shared is shared DSP channels and private is private DSP channels.

Table 182: LD 11 – Configure IP sets

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	l s c u	Terminal number for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
CUST	xx	Customer number, as defined in LD 15
BUID	<user_id>	Delible DN, Main Office User ID. Enter X to delete.
MOTN	l s c u	Main Office TN for Large System, Media Gateway 1000B and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
ZONE	<number>	Zone Number to which the IP Phone belongs.
CLS	VOLA VOLD	Allow/Deny Virtual Office operation from this TN.
	VOUA VOUD	Allow/Deny Virtual Office logon onto this TN using another phone (destination of Virtual Office logon).

Table 183: LD 15 – Choose zone (Current or Configured) to be used for zone-based digit manipulation feature

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE:	FTR_DATA	Features and options.
CUST	xx	Customer number.
VO_CUR_ZONE_ZDM	NO (default) YES	(Disable) enable using Current zone for zone-based digit manipulation.

Table 184: LD 15 – Choose zone (Current or Configured - default value) to be used for Time and Date

Prompt	Response	Description
REQ	CHG	Change existing data block
TYPE:	FTR_DATA	Features and options
CUST	xx	Customer number

Prompt	Response	Description
VO_CUR_ZONE_TD	NO (default) YES	(Disable) enable using Current zone for Time and Date.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 81: Backup and restore

Contents

This section contains information on the following topics:

Printing a backup rule on page 574

Introduction

The backup rules feature provides a mechanism for transferring the backup files produced by an Automatic Equipment Data Dump (EDD) operation to another location, such as an external File Transfer Protocol (FTP) server.

The backup rules group of commands was introduced in CS 1000 Release 4.0 with the single rule type: Secondary Call Server (SCS).

The backup rules feature applies to CP PIV system configurations, unless otherwise noted.

This feature does not actually perform an EDD, which must be initiated through an appropriate command in LD 43.

Backup rules

The backup rule types are as follows:

- FTP — an external FTP server is accessed for storing or retrieving backup data.
- FMD — fixed media device for storing backup data in a special directory on the /u partition. A CP PII system stores the data on a local hard drive, while a CP PIV uses an on-board CF card
- PFMD — fixed media device for storing backup data in a special directory on the /p partition.
- RMD — removable media device (faceplate CF card) for storing backup data. The RMD rule type is available only on a CP PM, CP PIV system.
- SCS — backup to the Secondary Call Server, the only backup rule type which can be referenced from Geographic Redundancy Database Replication Control (GRDRC) block.
- USB — removable USB device for storing backup data. The USB rule type is available only on CP DC, CP MG, and COTS systems.

You can define only 1 rule of type RMD, FMD, PFMD and USB for each Secondary Call Server IP address. You can define up to 50 rules of type SCS.

FTP rule type

The FTP rule type provides the ability to store backup files on an FTP server. This rule type defines the following six parameters:

- Name — ranges in length from 0 to 30 characters. All characters are supported with the exception of white space. Names do not have to be unique.
- IP address — specifies the FTP server IP address.
- Login name — specifies the FTP server login name. Ranges in length from 1 to 32 characters. All characters are supported with the exception of white space.
- Password — specifies the FTP server password. Ranges in length from 1 to 32 characters. All characters are supported with the exception of white space.
- Path — specifies the location of the file on the FTP server. Ranges in length from 1 to 64 characters. All characters are supported with the exception of white space.
- Number of versions — number of incremental backup versions preserved on the FTP server. Ranges from 1 to 10, with one as the default value. Note that this range differs from the SCS rule type, which is 2 to 10.

White space characters are not supported in backup rule parameters.

The Element Manager backup rule configuration page specific to the FTP rule type is shown in Figure 7: FTP rule type in Element Manager on page 556.

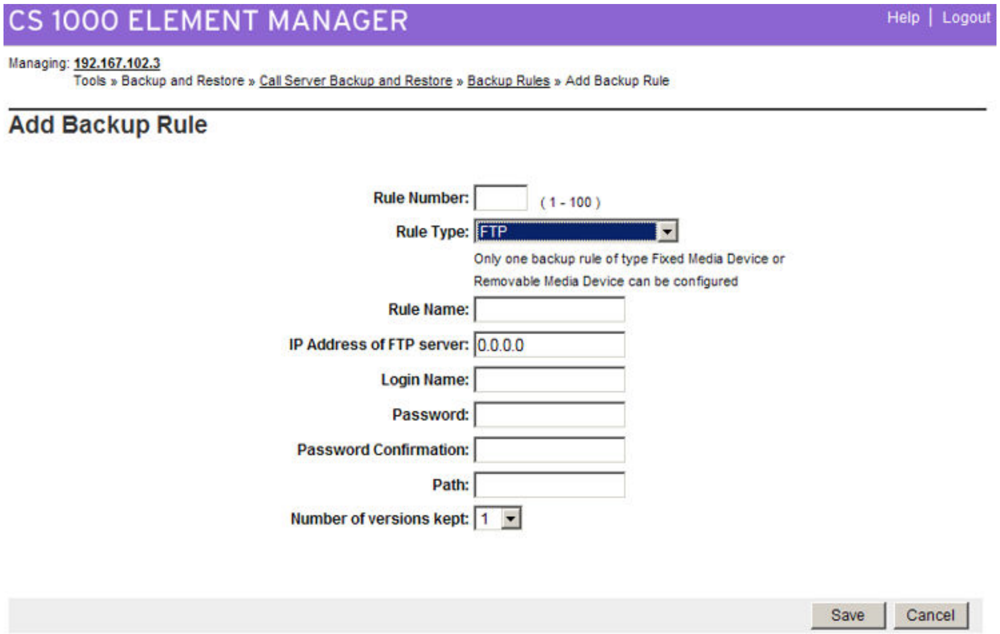


Figure 7: FTP rule type in Element Manager

The **backup rule number** control is static and cannot be changed by the user. Asterisk characters replace the actual password characters in the **password** and **password confirmation** controls.

The **Rule Name**, **Login name**, **Password**, **Password confirmation**, and **Path** controls restrict the maximum length of characters entered. An alert box appears when an invalid character (such as white space) is entered.

The text is as follows:

```
The input data contains one or more white space characters. White space characters
are not supported for this parameter.
```

Element Manager performs validation to ensure the content of the **Password** and **Password confirmation** controls boxes is identical. If it is not, these controls are cleared and an alert box appears with the following text:

```
The entered passwords do not match. Please try again.
```

The FTP rule type cannot be associated with a Geographic Redundancy Database Replication Control Block. All FTP backup rules must be excluded from the list of configured backup rules presented on the Element Manager Database replication control page.

FMD rule type

The FMD rule type provides the ability to store backup files on the local fixed media device (FMD). FMD is defined as the onboard Compact Flash (CF) card on a CP PIV system. The FMD rule type defines the following two parameters:

- Name: ranges in length from 0 to 30 characters. All characters are supported with the exception of white space. Backup rule names do not have to be unique.
- Number of versions: number of incremental backup versions preserved on the fixed media device. The range is from one to ten, with one as the default value. Note that this range differs from the SCS rule type, which is two to ten.

The Element Manager backup rule configuration page specific to the FMD rule type is shown in Figure 8: FMD rule type in Element Manager on page 557. The backup rule number control is static and cannot be changed by the user.

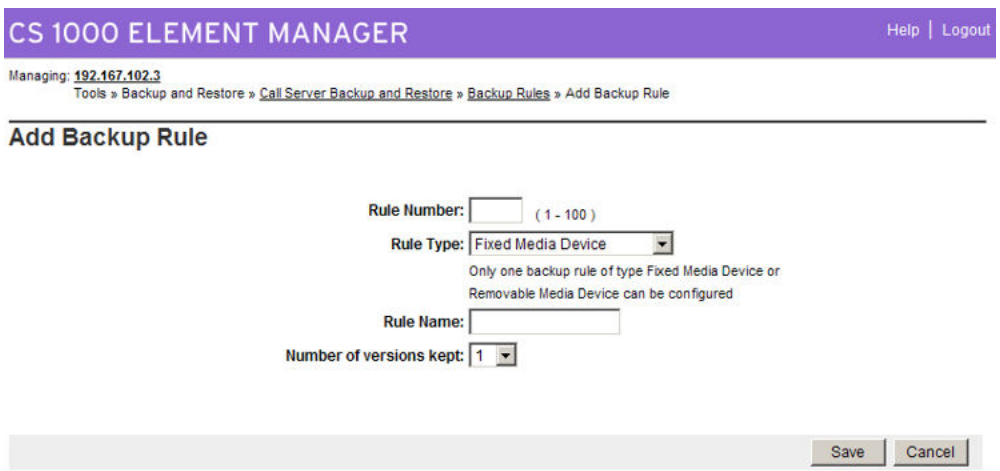


Figure 8: FMD rule type in Element Manager

The **rule name** control restricts the maximum length of characters entered, and also ensures no white space characters are used. An alert box is presented when an invalid character is entered.

The FMD rule type cannot be associated with a Geographic Redundancy Database Replication Control Block. All FMD backup rules must be excluded from the list of configured backup rules presented on the Element Manager Database replication control page.

PFMD rule type

The PFMD rule type provides the ability to store backup files on the local fixed media device (/p partition). PFMD is defined as the onboard Compact Flash (CF) card or Hard Disk drive (HDD) on all VxWorks, VxELLs CS platform systems. This rule is created by default and provides the ability to store backup data on the protected partition /p every week. If required, you can modify the PFMD rule definition and schedule.

If the Call Server system is a new installation, upgraded, or migrated to Communication Server 1000 Release 7.6 or higher, the PFMD backup rule is created by default as part of the conversion from the default or previous database to the Release 7.6 database. The PFMD rule uses the default backup schedule. All backup data files are stored each Saturday at 3:00 AM on the protected /p partition.

You can add, modify, or remove the PFMD rule manually using LD 117. The PFMD rule type defines the following two parameters:

- Name — ranges in length from 0 to 30 characters. All characters are supported with the exception of spaces. Backup rule names do not have to be unique.
- Number of versions — number of incremental backup versions. Ranges from 1 to 10, with 1 as the default value. (Note that this range differs from the SCS rule type, which is 2 to 10.)

The Element Manager backup rule configuration in PFMD rule type in Element Manager on page 558 is static and cannot be changed.

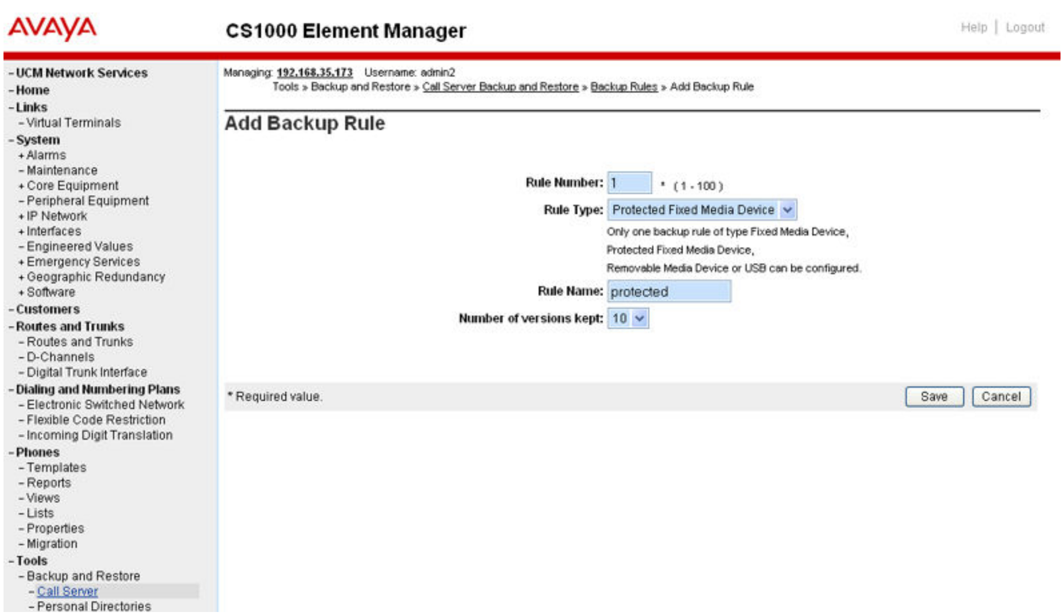


Figure 9: PFMD rule type in Element Manager

The PFMD rule type cannot be associated with a Geographic Redundancy Database Replication Control Block. All PFMD backup rules must be excluded from the configured backup rules listed on the Element Manager Database replication control page.

RMD rule type

The RMD rule type provides the ability to store backup files on the local RMD (faceplate CF card). The RMD rule type defines the following two parameters:

- Name: ranges in length from 0 to 30 characters. All characters are supported with the exception of white space. Backup rule names do not have to be unique.
- Number of versions: number of incremental backup versions preserved on the removable media device. The range is from one to ten, with one as the default value. Note that this range differs from the SCS rule type, which is two to ten.

The Element Manager backup rule configuration page specific to the RMD rule type is shown in Figure 10: RMD rule type in Element Manager on page 559. The **backup rule number** control is static and cannot be changed by the user.

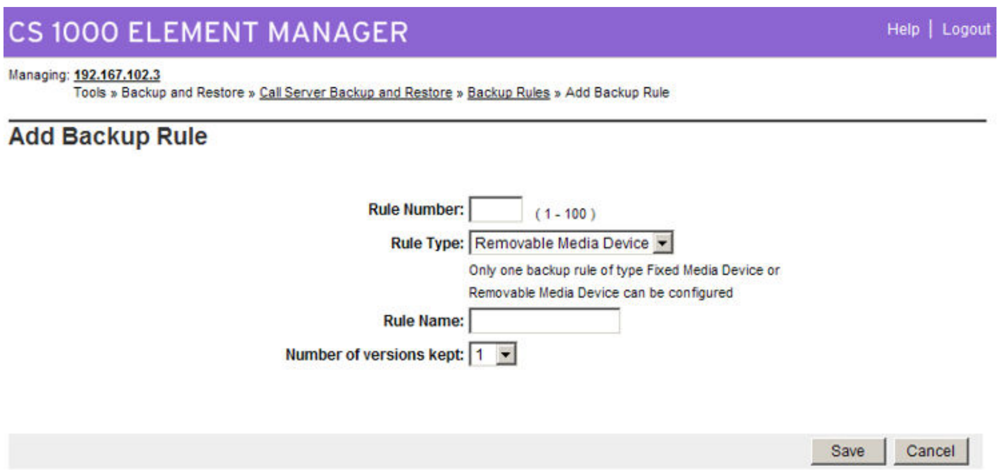


Figure 10: RMD rule type in Element Manager

The **rule name** control restricts the maximum length of characters entered, and also ensures no white space characters are used. An alert box is presented when an invalid character is entered.

The RMD rule type cannot be associated with a Geographic Redundancy Database Replication Control Block. All RMD backup rules must be excluded from the list of configured backup rules presented on the Element Manager Database replication control page.

SCS rule type

CS 1000M Large Systems (CPP) and Avaya Communication Server CS 1000E systems both provide redundancy using dual processors. This allows a system to remain operational following a local component failure.

Geographic Redundancy further increases the reliability of CS 1000M Large Systems (CPP) and Avaya Communication Server CS 1000E systems by providing a remote system to serve as a backup for a local system. The remote backup ensures continued service for all IP Phones in case of a catastrophic failure (for example, as a result of floods or fire).

The secondary system database is replicated from the primary system database. As a result, the secondary system must be the same type of system as the primary system. That is, a CS 1000M Large System can be backed up only by another CS 1000M Large System and a CS 1000E system can be backed up only by another CS 1000E system.

To perform the database-replication process, an SCS backup rule must be defined on the primary system. The backup rule identifies the destination ELAN network interface IP address on the secondary system for the database replication. It also defines the number of versions of the database that are kept on the secondary system.

To complete the database replication successfully, the Backup Rule must be referenced in the Database Replication Control Block. For more information, see *Avaya System Redundancy Fundamentals* (NN43001-507).

The secondary system also uses the Backup Rule during the database-restore operation to identify the appropriate database to restore.

A backup rule can also be defined on the secondary system to provide replication to the primary system after a long-term failure. For more information, see *Avaya System Redundancy Fundamentals* (NN43001507).

The Element Manager backup rule configuration page specific to the SCS rule type is shown in Figure 11: SCS rule type in Element Manager on page 560. The **backup rule number** control is static and cannot be changed by the user.

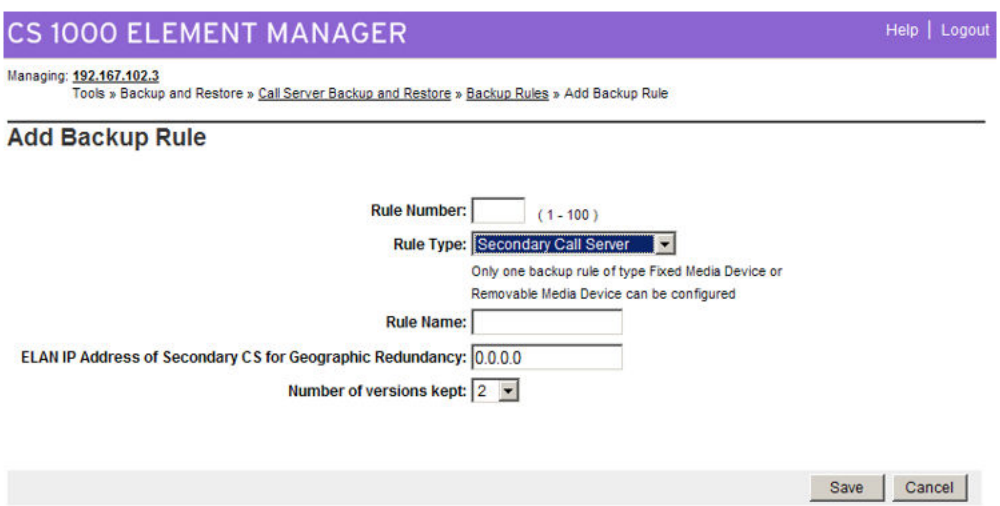


Figure 11: SCS rule type in Element Manager

USB rule type

The USB rule type provides the ability to store backup files on a removable USB media device. USB is defined as the removable USB media device on a CP DC, CP MG, or COTS Linux Call Server systems.

The USB rule type defines the following two parameters:

- Rule Name — can be 0 to 30 characters. All characters are supported with the exception of spaces. Backup rule names do not have to be unique. An alert box displays if an invalid character is entered.
- Number of versions — number of incremental backup versions. Ranges from 1 to 10, with 1 as the default value. (Note that this range differs from the SCS rule type, which is 2 to 10.)

The **Element Manager** backup rule configuration is static and cannot be changed by the user.

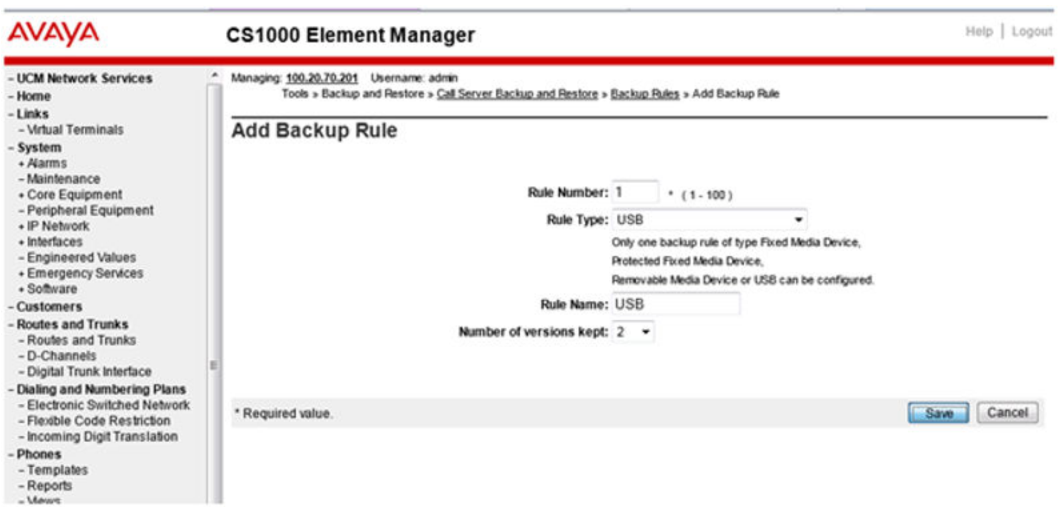


Figure 12: USB rule type in Element Manager

The USB rule type cannot be associated with a Geographic Redundancy Database Replication Control Block. All USB backup rules must be excluded from the configured backup rules listed on the Element Manager Database replication control page.

Backup schedules

Backup schedules provide the user with the ability to schedule backup operations associated with a specified backup rule. Each backup schedule defines a total of five associated parameters, as follows:

- Schedule number: up to ten backup schedules can be defined, numbered from one to ten.
- Rule for BKUP: specifies the backup rule number associated with this backup schedule. The backup rule number must be previously configured.
- FREQ: defines how often the scheduled backup operation occurs. The default is D. Not more than one backup schedule can be defined with FREQ set to the value A. Options are:
    - M (monthly)
    - W (weekly)
    - D (daily)
    - A (automatic — immediately after every EDD).
- DAY: specifies the day on which the backup occurs. with a default value of SU. When FREQ is M, the range is 1 to 31 with a default value of 1. This parameter does not apply when FREQ is set to either of the values D or A. When FREQ is W, the range is the days of the week as follows:
    - SU
    - MO
    - TU
    - WE
    - TH
    - FR
    - SA
- HOUR: specifies the hour in the day at which the backup occurs. The range is 0 to 23, with a default of 3. This parameter does not apply when FREQ is set to the value A.

Backup schedules are only supported on CP PIV systems. A backup schedule can be created, modified, deleted, and printed by the respective command options **NEW**, **CHG**, **OUT**, and **PRT**.

Element Manager backup schedules page

The Element Manager backup schedules page allows the user to edit or delete all currently configured schedules or add a new backup schedule. Element Manager ensures all schedule information is kept current by querying the call server for all configured backup schedules and backup rules. The backup schedules page is illustrated in Figure 13: Backup schedules in Element Manager on page 563.

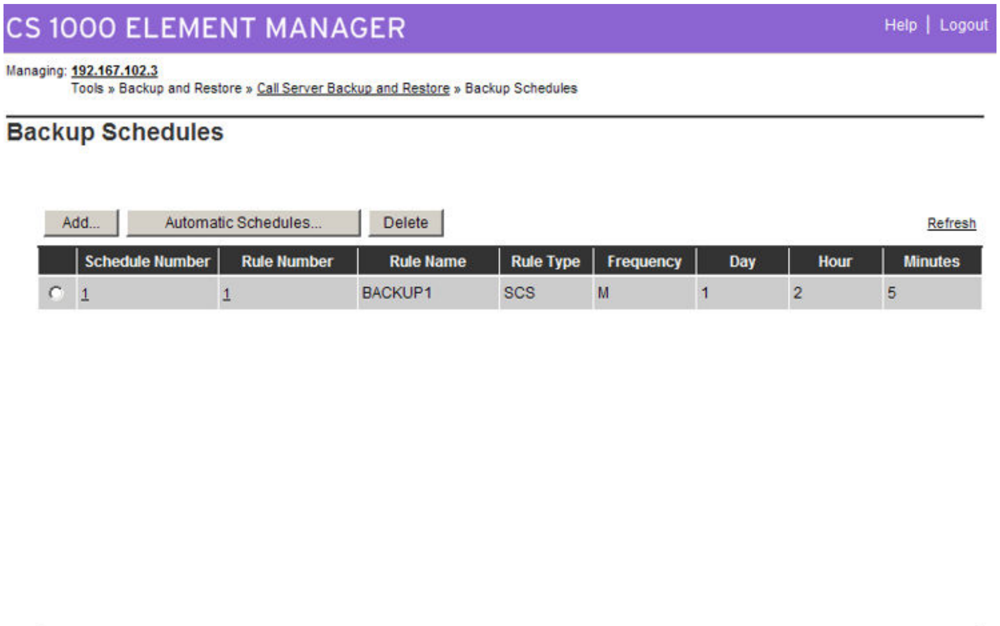


Figure 13: Backup schedules in Element Manager

When the user selects a backup schedule to add and selects **Add**, or alternatively selects one of the **Edit** buttons, the Element Manager returns to the backup schedule configuration page.

The drop-down list specifying which backup schedule number to add only lists those backup schedules which were not previously configured on the call server.

If one or more backup rules are defined, then the backup schedules page continues to load. If no backup rules are defined, the user is presented with a confirmation box containing OK and Cancel buttons along with the following text:

```
At least one Backup Rule must be defined prior to using this option.
Click the OK button to begin creating one.
```

Selecting **OK** redirects Element Manager to the backup rules page. Selecting **Cancel** returns the browser to the backup and restore page.

Each listed backup schedule's **Delete** button provides the ability to delete the corresponding backup schedule without accessing the corresponding backup schedule configuration page. To confirm the delete, click OK. (If the confirmation is declined, the user is returned to the backup schedules page).

```
Click OK to confirm delete.
```

Element Manager sends a request to the call server to delete the specified backup schedule. Upon successful completion of a delete operation, the Element Manager browser reloads the backup schedules page. The deleted schedule is no longer displayed. If the delete operation cannot be completed due to an error, an alert box appears, followed by a return to the backup schedules page.

Element Manager backup schedule configuration page

The Element Manager backup schedule configuration page provides the ability to specify the settings of a particular backup schedule.

The Element Manager backup schedule configuration page is illustrated in Figure 14: Backup schedule configuration in Element Manager on page 564.

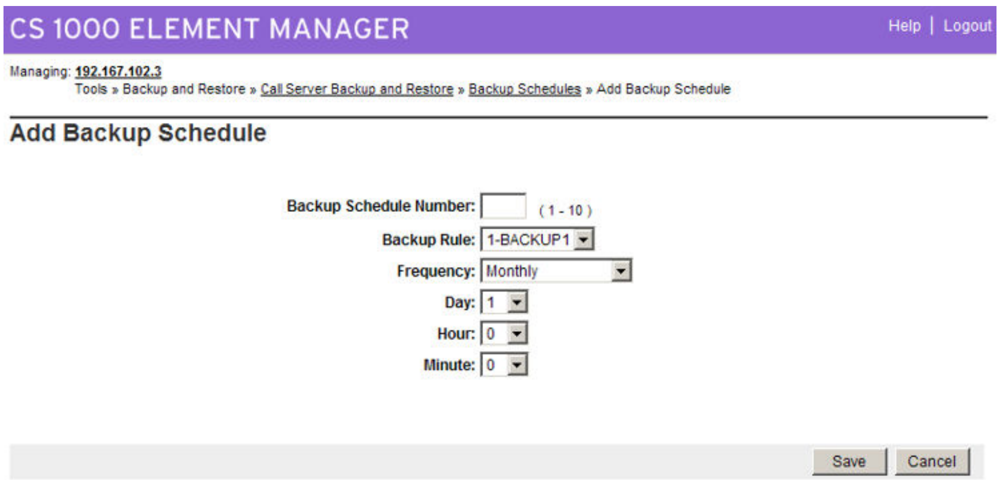


Figure 14: Backup schedule configuration in Element Manager

The backup schedule number control is static and is set based on the selections made in the backup schedules page. Possible values are one through ten.

The backup rule selection control is dynamically updated with the list of configured backup rules on the call server, with possible values being 1 through 100. The default, when adding a new backup schedule, is the lowest numbered configured backup rule. The backup rule name and backup rule type controls are read-only and are dynamically filled in, based on the selection of the backup rule number selection control.

The frequency selection control contains the selection options of:

- Monthly
- Weekly
- Daily
- Triggered by EDD

Note that the Triggered by EDD option corresponds to the overlay option `Automatic` .

The option list for the day selection control is dynamically set based on the selected value for the frequency selection control. When frequency is set to **Monthly**, then the day option list consists of the range of numbers 1 to 31, with a default of 1. When frequency is set to **Weekly**, then the day option list consists of the days of the week as follows:

- Sunday
- Monday
- Tuesday
- Wednesday
- Thursday
- Friday
- Saturday

The default is **Sunday**. When frequency is set to daily or **Triggered by EDD**, then the setting of the day selection control does not apply and is disabled.

When the user attempts to select the **Triggered by EDD** option for frequency, Element Manager ensures there are no other backup schedules which have this parameter set to **Triggered by EDD**. If this is detected, the frequency selection control is returned to its previous setting and an alert box containing the following text appears:

```
No more than one Backup Schedule can have a Frequency selection of "Triggered by
EDD". Please select another option.
```

The hour selection control contains the options 0 through 23. When frequency is set to **Triggered by EDD**, the setting of the hour selection control does not apply and is disabled.

Selecting the **Submit** button initiates transfer of the backup schedule configuration to the call server.

Upon successful completion of a submit operation, the Element Manager browser returns to the backup schedules page. If the operation cannot be completed due to an error, an alert box appears, followed by a return to the backup schedules page.

Selecting the **Refresh** button causes Element Manager to reload the backup schedule configuration page with the latest data from the call server for the backup schedule. Upon successful completion of a refresh operation, the Element Manager browser returns to the backup schedules page. If the operation cannot be completed due to an error, an alert box appears, followed by a return to the backup schedules page.

Selecting the **Delete** button opens a confirmation box with the following text:

```
Click OK to confirm delete.
```

If the delete is confirmed, Element Manager sends a request to the call server to delete the specified backup schedule. Upon successful completion of a delete operation, the Element Manager browser returns to the backup schedules page. If the operation cannot be completed due to an error, an alert box appears, followed by a return to the backup schedules page.

If the user declines the confirmation, the Element Manager browser returns to the backup schedules page.

Note that when the backup schedules configuration page is loaded as a result of the user selecting the **Add** button on the backup schedules page, both the **Refresh** and **Delete** buttons are disabled.

Element Manager ensures the information in this page is kept current by querying the call server for the requested backup schedule data and the currently configured backup rules.

Backup scheduler

Backup operations can be scheduled to run automatically at user-defined intervals to user-defined destinations (such as external FTP servers or removable CF cards). This is done by defining Backup Schedules in LD 117.

The following conditions apply:

- For each schedule, the frequency (monthly, weekly, daily or automatically after every EDD) must be defined along with the day and time of the activation. A maximum of ten schedules can be defined.
- For each schedule, a backup rule must be used. It can be any rule type: FTP, FMD, RMD, or SCS.

This feature is not applicable to the Secondary Call Server of the Geographic Redundancy feature.

Backup and restore maintenance

Corrupted backup data

Backup data is protected in the following ways:

- Traditional protection in the form of an archive.dat file, including the list and checksum of the files backed up.
- The new backup operation, which "envelopes" the backup data by using tar and gzip utilities to provide additional checksum protection.

If backup data is corrupted, the restore operation is aborted and the proper alarm report is generated.

FTP data transfer failure

The FTP client facility is used to access a remote FTP server for the following operations:

- manual or scheduled backup
- restore according to backup rule (manual activation only)
- viewing the backup data versions according to backup rule (manual activation only)

Manual operation

FTP connection failure on a manual operation causes generation of the proper SRPT system, providing the details of the failure (FTP server unreachable, login name/password rejected, requested directory path not found, or data transfer failed). The operation is aborted.

Automatic operation

FTP connection failure on automatic operation is automatically attempted again for a hard-coded number of times (five) with a hard-coded delay (ten seconds) between attempts.

An automatic operation FTP session occurs in sequential steps, such as copying the file to the remote site, deleting and renaming operations for the multi-version support, and so on. Any attempt to connect continues from the step that failed during the previous attempt to connect. If connection attempts exceed five, the proper SRPT system alarm report is generated and the backup operation is aborted.

Defining backup schedules

- In LD 117, define a new backup rule (or rules) by using **`NEW/CHG/OUT/PRT BKPR`** commands.
- In LD 43, verify the newly defined backup rules by performing manual backup (**`BKR R`**), printing backup data versions (**`DAT R, DAT R V`**), and manual restore (**`RSR R`**).
- In LD 117, define a new backup schedule (or schedules) with reference to the newly defined and verified backup rules (**`NEW/CHG/OUT/PRT BKPS`**).
- Wait for the scheduled time to occur.

Backup operations performed manually by entering the **`BKR`** command in LD 43 (or configured in backup scheduler) provide multiple versions of the backup data preserved on the storage device. These multiple versions use incremental numbering.

When performing a manual restore by entering the RSR command in LD 43, the version of the backup data must be entered as a parameter. In most cases, the latest version is chosen as version number one, but under certain conditions, information about all stored backup versions may be required. This query is provided by the enhanced **DAT** (data versions) command in LD 43.

The DAT command provides two possible outputs:

- A list of backup versions including incremental version numbers and date and time stamps of each version created. The **DAT** command must be entered with the backup rule number specified.
- Detailed info for a specified incremental backup version that includes the software issue, size in records, and sequential number of the DUMP operation. The **DAT** command must be entered with the backup rule number and incremental backup version number specified.

The modified Element Manager call server restore page is shown in Figure 15: Modified call server restore page in Element Manager on page 568. The new backup data versions option **DAT [R [V]])** is appended to the existing list of options in the action control. This option is displayed only for CP PIV call servers.

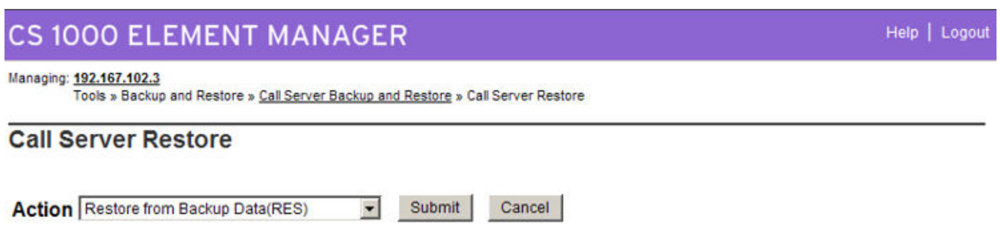


Figure 15: Modified call server restore page in Element Manager

When the backup data versions **(DAT [R [V]])** option is selected, two additional drop-down list controls are displayed to query the user for the backup rule number, and optionally the backup version associated with the selected backup rule number.

The **backup rule number** control provides a dynamic list of all currently configured backup rules. The default setting is the lowest number backup rule configured. This functionality is exactly the same as the **backup rule number** control associated with the restore according to rule (RSR X Y) option in the action control.

The **backup data versions** control provides a list of backup data versions associated with the selected backup rule number. The option list is based on the value of the versions kept parameter of the currently selected backup rule number. This functionality is the same as the **restore version** control associated with the restore according to rule (RSR X Y) option in the action control with the exception that the additional selection option not specified is included, since this parameter is optional. The default for this control is the not specified option.

Selecting the **Submit** button results in the command being submitted to the call server for processing. The response from the call server is displayed verbatim in the call server restore page.

If no backup rules are configured, Element Manager displays an alert box with the following text.

```
There are no Backup Rules currently configured. At least one Backup Rule must be
configured in order to use this option.
```

Selecting the **OK** button restores the action control to the previously selected setting.

Backup history

The Backup History (BKH) command in LD 43 provides a history of attempts at archiving backup data using configured backup rules. The call server stores up to 100 previous attempts spanning all configured backup rules. The backup history command provides the following options:

- List backup attempts spanning all configured backup rules. The command provides the option to limit the number displayed, up to a maximum of 100.
- List backup attempts associated with a specific backup rule. The command provides the option to limit the number displayed, up to a maximum of 100.

Access to the data provided by the **BKH** command is incorporated into Element Manager on the backup rules page. The **Backup History** button is next to the **Edit** button for each backup rule, as shown in Figure 16: Backup history page in Element Manager on page 569.

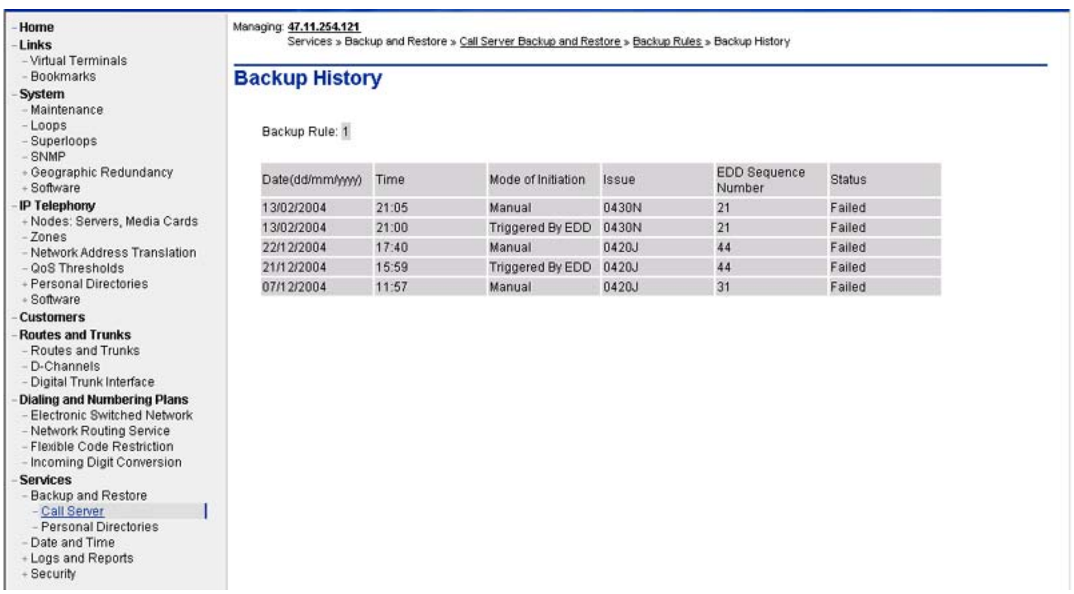


Figure 16: Backup history page in Element Manager

Clicking the **Backup History** button redirects Element Manager to the backup history page, which lists all recorded backup attempts (potentially up to 100) for the specified backup rule, in reverse chronological order.

Backup history on the Element Manager Home page

The Element Manager Home page provides a brief summary of the status of backup data archives, as shown in Backup history on the Element Manager Home page. This summary is only applicable to CP PII and CP PIV systems.

The information required to populate this section of the Element Manager Home page is retrieved from the call server using the BKH command. Element Manager must parse the retrieved information to determine the details of the last successful backup archive.

The **Last Successful Backup Archive** field and the corresponding **Backup Archive Initiation** field are displayed only when the value of the **Status** field is **Failed**.

If that information is not available for the last backup archive and/or the last successful backup archive, the corresponding fields are assigned the text `Not Available` .

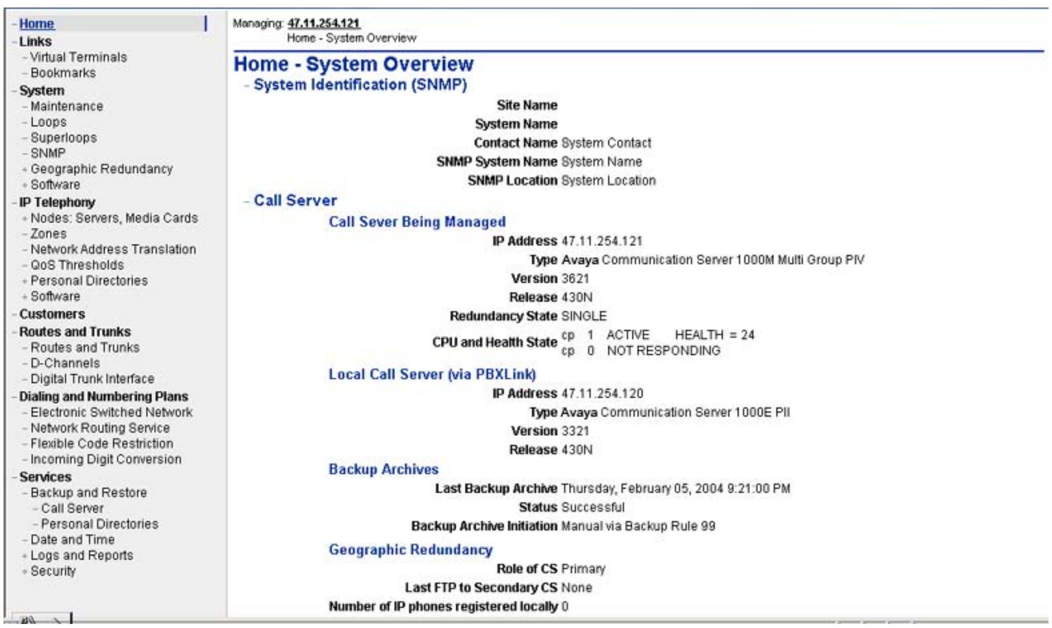


Figure 17: Backup history on the Element Manager Home page

The reliability of data displayed on the Element Manager Home page is dependent on the time at which the user logged on to Element Manager. If information changes following login, it is not reflected on the Element Manager Home page during the current Element Manager logon session. To redisplay the most current information, the user must open a new Element Manager logon session, either by logging out of the current session and logging back in, or by logging on using another User ID.

Overlay administration

Backup to an external FTP server (FTP backup rule)

During a backup operation, the backup data files are grouped into one file using the tar utility, compressed using the gzip utility, and copied to the local hard disk.

During a restore operation, the specified version of the backup data file is copied from the local hard disk, de-grouped using the tar utility, and decompressed using the gzip utility.

Table 185: LD 117 Ethernet and Alarm Management

Command	Description
NEW/CHG BKPR <rule number 1-100> FTP <IP addr> <login> <pwd> <path> [<N of versions 1-10>] [<name>]	Add or change a backup rule, where:
	rule number = 1-100. Up to 100 rules can be defined. Each rule is a pattern that can be further used. These rules can be used by the GRDRC block, defined in LD 117, by the backup schedules, or for manual backup and restore operation (BKR/RSR commands activated from LD 43).
	FTP = mnemonic for this rule type
	IP addr = IP address of the FTP server to be accessed for storing (Backup) or retrieving (Restore) backup data
	login = login name to access the FTP server, up to 32 characters
	pwd = login password to access the FTP server, up to 32 characters
	path = path on the FTP server where the backup data file (or files for incremental versions) is located, up to 64 characters
	N of versions = (1)-10 number of incremental backup data versions preserved on the FTP server
	name = rule name - text without white spaces, up to 30 characters
The only backup rule type that can be referenced from GRDRC is SCS. The <name> parameter is also added as optional when defining a new backup rule with SCS type (introduced in CS 1000 Release 4.5Geographic Redundancy).	

Backup to FMD (FMD backup rule)

During a backup operation, the backup data files are grouped into one file using the tar utility, compressed using the gzip utility, and copied to the FMD.

During a restore operation, the specified version of the backup data file is copied from the FMD, de-grouped using the tar utility, and decompressed using the gzip utility.

Table 186: LD 117 Ethernet and Alarm Management

Command	Description
NEW/CHG BKPR <rule number 1-100> FMD [<N of versions>] [<name>]	Add or change a backup rule, where:
	rule number = 1 - 100. Up to 100 rules can be defined. Each rule is a pattern that can be further used. RMD rules can be used by the backup schedules or for manual backup and restore operation (BKR/RSR commands activated from LD 43).
	FMD = mnemonic for this rule type
	N of versions = (1)-10 number of incremental backup data versions preserved on the local hard disk
	name = rule name - text without white spaces, up to 30 characters
The <name> parameter is also added as optional when defining a new backup rule with SCS type (introduced in CS 1000 Release 4.5Geographic Redundancy).	

Backup to RMD (RMD backup rule)

During a backup operation, the backup data files are grouped into one file using the tar utility, compressed using the gzip utility, and copied to the local RMD.

During a restore operation, the specified version of the backup data file is copied from the local RMD, de-grouped using the tar utility, and decompressed using the gzip utility

Table 187: LD 117 Ethernet and Alarm Management

Command	Description
NEW/CHG BKPR <rule number 1-100> RMD [<N of versions>] [<name>]	Add or change a backup rule, where:
	rule number = 1 - 100. Up to 100 rules can be defined. Each rule is a pattern that can be further used. RMD rules can be used

Command	Description
	by the backup schedules or for manual backup and restore operation (BKR/RSR commands activated from LD 43). RMD = mnemonic for this rule type N of versions = (1)-10 number of incremental backup data versions preserved on the local removable media device name = rule name - text without white spaces, up to 30 characters
The <name> parameter is also added as optional when defining a new backup rule with SCS type (introduced in CS 1000 Release 4.5Geographic Redundancy).	

Backup to SCS (SCS backup rule)

During a backup operation, the backup data files are grouped into one file using the tar utility, compressed using the gzip utility, and copied to the FMD (CP PIV).

During a restore operation, the specified version of the backup data file is copied from the FMD (CP PIV), de-grouped using the tar utility, and decompressed using the gzip utility

Table 188: LD 117 - Ethernet and Alarm Management

Command	Description
NEW BKPR xxx aaa b...b yy	Add a new Backup Rule, where: • xxx = Backup Rule number ID = 1-100. • aaa = rule type = SCS. Currently, this is the only rule type that exists: it allows direct replication to another system. • b...b = ELAN network interface IP address of the destination system. • yy = the number of database versions to save on the destination system = (2)-10.
CHG BKPR xxx aaa b...b yy	Change a Backup Rule, where: • xxx = Backup Rule number ID = 1-100. • aaa = rule type = SCS. Currently, this is the only rule type that exists: it allows direct replication to another system. • b...b = ELAN network interface IP address of the destination system. • yy = the number of database versions to save on the destination system = (2)-10.

Command	Description
OUT BKPR xxx	Remove backup rule, where: • xxx = Backup Rule number ID = 1-100
PRT BKPR xxx	Print backup rule, where: • xxx = Backup Rule number ID = 1-100 If no rule number is entered, then all Backup Rules are printed.

Removing a backup rule

Table 189: LD 117 Ethernet and Alarm Management

Command	Description
OUT BKPR <rule number 1-100>	Remove a backup rule
Backup rules must be entered one per command line. Entering multiple backup rules on one command line is not supported.	

Printing a backup rule

Table 190: LD 117 Ethernet and Alarm Management

Command	Description
PRT BKPR <rule number 1-100> (ALL)	Print a backup rule

```
                    number of Backup Rules defined = 4
```

```
NN            Dest          Parameters          N of V         Name
```

```
1            SCS          111.22.133.44        10             1+1_GR_DB_dup
```

```
2             FTP           55.166.77.188        3
guest
switch1/backup
```

```
3             FMD                                  2
```

Command	Description
`4          RMD`	`4          To_CF_4_vers`
For security reasons the FTP password is not printed.	

Adding or changing a backup schedule

Table 191: LD 117 Ethernet and Alarm Management

Command	Description
NEW/CHG BKPS <schedule number 1-10> <Rule for BKUP> <FREQ> <DAY> <HOUR>	Add or change a backup schedule, where:
	Rule for BKUP = number of the backup rule for the scheduled backup operation
	FREQ = M/W/(D)/A - defines how often the scheduled backup should take place: M - monthly W - weekly D - daily A - automatically immediately after every EDD operation activated. There cannot be more than 1 schedule defined with FREQ = A
	DAY = day of the week (SU) / MO / TU / WE / TH / FR / SA if FREQ = W or date of the month (1) - 31 if FREQ = M If FREQ = M and the day specified is greater than the number of days in the current month, the backup will take place on the last day of the current month If FREQ = D, the next parameter will be HOUR and not DAY
	HOUR = 0- (3) -23 - hour of the day
This rule type is not allowed if the GRPRIM/GRSEC package is equipped and the rule is used in GRDRC.	

Removing a backup schedule

Table 192: LD 117 Ethernet and Alarm Management

Command	Description
OUT BKPS <schedule number 1-10>	Remove a backup schedule

Printing a backup schedule

Table 193: LD 117 Ethernet and Alarm Management

Command	Description
PRT BKPS <schedule number 1-10> /ALL	Print a backup schedule

```
number of Backup Schedules defined = 3
NN    Backup_Rule    Frequency    Day    Hour
1     2              Monthly      1      3
2     3              Weekly       Su     5
3     4              Auto
```

Chapter 82: Boss/Secretary Filtering Enhancement

Contents

This section contains information on the following topics:

Feature description

The Boss/Secretary Filtering Enhancement (BSFE) feature is designed for a boss/secretary environment.

Prior to the introduction of the BSFE feature, a boss could forward incoming calls to secretary/secretaries for screening.

With the BSFE feature, incoming calls are forwarded from the boss to a designated secretary using the Call Forward and Busy Status (BFS) key. A maximum of 16 BFS keys can be configured on the boss telephone. A corresponding BFS key is configured on each secretary telephone. The following enhancements are also introduced by this feature:

- Display capabilities: If the Display key is pressed during an incoming filtered call, the calling party's name and number appear on the telephone display.
- Transfer capabilities: If a secretary presses the BFS key once, listens for the boss to pickup and presses the BFS key a second time, the incoming filtered call is transferred back to the boss.

- New Classes of Service: The Boss Secretary Filtering Enhancement Class of Service Allowed (BFEA) or Denied and the Recall to Boss Allowed (RCBA) or Denied (RCBD).
- Key Lamp status: The BSFE feature allows configuration of the LCD indicator for the BFS key. It is possible to configure the same LCD lamp status to
    - Dark (key lamp is off)
    - Lit (key lamp is steadily lit)
    - Wink
    - Flash (continual flash of light, 60 ipm)

The BSFE feature is configured on the boss telephone, with a defined BFS key for each secretary that the boss may select to filter the boss' incoming calls. The telephone will also have a designated key matching the boss key. The BFS key must be a single appearance DN for the boss and the secretary telephones. The BFS keys for the boss/secretary telephones are configured in pairs and are on the same node.

Operating parameters

Proprietary telephones with display support the BSFE feature. The BSFE feature cannot be configured for analog (500/2500) telephones or Integrated Services Digital Network (ISDN) BRI telephones. The ringing appearance of the DN can be on an analog (500/2500) telephone but not for a private line.

The BSFE feature cannot be activated simultaneously with the following features:

- Call Forward and Busy Status
- Call Forward All Calls
- Remote Call Forward
- Flexible Feature Code Boss Secretarial Filtering

The BSFE feature supports a maximum of 16 secretary telephones associated with the boss telephone.

With the BSFE feature, the BFS key of the boss is generally non-ringing with key lamp indication notification; the secretary telephone is set up as ringing.

The BSFE feature cannot be activated if the DN of either telephone is configured as an Automated Call Distribution (ACD) key.

Feature interactions

Hold

If the BSFE feature is active, the secretary answers the incoming boss call by pressing the SCR key or by pressing the BFS key. If the call is answered on the BFS key, pressing the key a second time will automatically put the call on hold and autodial the DN of the boss. If the class of service of the telephone is Auto Hold Allowed (AHA) and the call is on the BFS key, pressing the SCR key a second time puts the call on hold. If the class of service of the telephone is Auto Hold Denied (AHD) and the call is on the BFS key, pressing the SCR key again releases the call.

Hotline

Hotline takes precedence over BSFE. Hotline calls to the boss telephone are not filtered, even if the BSFE feature is active. The hotline calls are directed to the boss telephone.

Voice Call

If the Voice Call key/lamp is configured as the boss DN on a third party's telephone, the call is not filtered by the BSFE feature and the call terminates on the boss telephone.

Voice Mail

If a call is unanswered, whether the BSFE feature is active or deactivated, the voice mail message is sent directly to the voice mail box of the boss.

The BSFE feature takes precedence over the following features:

- Camp On If the BSFE feature is active on the boss telephone, the incoming calls are not camped on this DN but are sent directly to the secretary telephone.
- Call Waiting If a call comes in while the boss is on a call and the BSFE feature is active, the call is sent directly to the secretary telephone.
- Call Forward and Hunt Override If a secretary calls the boss without using the Call Forward and Busy Status (BFS) key, the call goes back to the secretary. If the secretary uses the BFS key when calling the boss, the call goes to the primary DN of the boss.

- Do Not Disturb If the BSFE feature is active on the boss telephone, the Do Not Disturb (DND) is overridden and the call is sent directly to the secretary.
- Hunting If the boss has Hunt configured and the BSFE feature is active, an incoming call is forwarded to the secretary, not sent through the hunt chain. If the secretary telephone is busy, the call follows the secretary hunt list.
- Make Set Busy If the BSFE feature and the MSB key is active, the incoming call is sent directly to the secretary; the caller does not receive a busy tone.
- Private Line Private Line calls are filtered by the secretary if the BSFE feature is active.

Feature packaging

This feature is included in base system software.

Feature implementation

Important:

The technician must be aware of the various configurations allowed for the LCD lamp notification states (dark, lit, wink, and flash) to avoid user confusion. The default lamp status states are shown below.

Boss telephone	Boss telephone with BFS deactivated	Boss telephone with BFS activated
Idle Busy	Dark Lit	Wink Flash

Table 194: LD 15 - Configure the lamp status for the Boss/Secretary Filtering Enhancement feature.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	FTR	Features and Options data.
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.

Prompt	Response	Description
BSFE	YES	YES = Allow Boss/Secretary Filtering Enhancement feature. (NO) = Deny Boss/Secretary Filtering Enhancement feature.
- ACT_IDLE	(WINK) FLSH LIT DARK	Boss's Lamp status when BSFE is active and telephone is idle. LCD Lamp flash rate is 60 impulses per minute. LCD Lamp flash rate is 30 impulses per minute. LCD Lamp is on. LCD Lamp is dark.
- ACT_BUSY	(FLSH) WINK LIT DARK	Boss's Lamp status when BSFE is active and telephone is busy. LCD Lamp flash rate is 30 impulses per minute. LCD Lamp flash rate is 60 impulses per minute. LCD Lamp is on. LCD Lamp is dark.
- DACT_IDLE	(DARK) WINK LIT FLSH	Boss's Lamp status when BSFE is disabled and telephone is idle. LCD Lamp is dark. LCD Lamp flash rate is 60 impulses per minute. LCD Lamp is on. LCD Lamp flash rate is 30 impulses per minute.
- DACT_BUSY	(LIT) WINK FLSH DARK	Boss's Lamp status when BSFE is disabled and telephone is busy. LCD Lamp is on. LCD Lamp flash rate is 60 impulses per minute. LCD Lamp flash rate is 30 impulses per minute. LCD Lamp is dark.

Table 195: LD 15 - Configure Offhook Alarm Security.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	OAS	Off-Hook Alarm Security options.
CUST		Customer number
	0-99	Range for Large System and Avaya CS 1000E system.
ODN0	xxxx	Offhook Alarm Security for zone 0

Table 196: LD 11 - Configure the Boss/Secretary Filtering Enhancement feature for Meridian proprietary telephones.

Prompt	Response	Description
REQ:	NEW CHG	Add new data. Change existing data.
TYPE:	a...a	Telephone type, as defined in LD 11.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DES	x..x	Office Data Administration System Designator
CUST	xx	Customer number, as defined in LD 15
...	...	

Prompt	Response	Description
CLS	BFEA	BFEA = Allow Boss/Secretary Filtering Enhancement for telephone. (BFED) = Deny Boss/Secretary Filtering Enhancement for telephone.
CLS	RCBA	RCBA = Allow Recall to boss on telephone basis. (RCBD) = Deny Recall to boss on telephone basis. This class of service forwards unanswered calls back to the boss after a specified number of rings.
...	...	
KEY	xx BFS l s c u	Call Forward and Busy Status (BFS) key. xx = key number. l = loop, s = shelf, c = card, u = unit for Large Systems. The TN can be the same telephone or any other digital telephone in the same node. Configure the TN of the same telephone against the BFS key only if the Class Of Service is BFEA.

Feature operation

To control the BSFE feature from the boss telephone:

Activate:

1. Press the BFS boss key once. The display shows:

   `PRESS BFS KEY OF SEC.`

2. Press the specific BFS secretary key to designate the secretary to filter the calls. The designated secretary's BFS key lamp winks on all telephones with the default lamp status.

The display clears on the boss telephone. To refresh the display, press the release key.

Deactivate:

1. Press the BFS boss key once. The display shows:

CANCEL FILTERING?

2. Press the BFS boss key for the second time. The feature is deactivated. The designated boss BFS key lamp turns DARK on all telephones with the default lamp status.

To control the BSFE feature from the secretary telephone:

Activate:

1. Press the BFS boss key once The display shows

   ACTIVATE FILTERING?

2. Press BFS boss key for the second time. This telephone becomes the secretary telephone. The display is cleared. The designated boss BFS key lamp WINKS on all telephones with the default lamp status.

Deactivate:

1. Press BFS boss key for once. The display shows:

   CANCEL FILTERING?

2. Press the BFS boss key second time. The feature is deactivated. The display is cleared. The designated boss BFS key lamp turns DARK on all telephones with the default lamp status.

To modify the BSFE from another secretary telephone:

1. Press the boss BFS key from another secretary telephone once. The display shows:

   MODIFY FILTERING?

2. Press the boss BFS key from the same telephone the second time. This secretary telephone becomes the new secretary filtering the calls of the boss telephone. The display is cleared.

Accept incoming call by boss:

1. Go offhook; press the SCR key.
2. Press the BFS boss key.

To transfer an incoming call from the secretary to the boss telephone:

1. Go off hook/press the SCR key to answer the ringing call.
2. Press the BFS boss key for the first time. The boss telephone rings.
3. The boss telephone answers the call.
4. Press the BFS boss key for the second time; this moves the call from the secretary telephone to the boss telephone.

The display - boss and secretary:

1. Press the Display key.
2. Press the BFS key. The telephone display shows the DN number of telephone filtering the boss calls.
3. The name and number of calls being filtered is displayed on the boss telephone.

   When the BSFE feature is activated on the boss telephone, the BFS key flashes on all secretary telephones associated with the boss telephone. This indicates the boss calls are being filtered. Each secretary can press the BFS key to display on their telephone. The secretary telephone filters the calls.

Chapter 83: Bridging

Contents

This section contains information on the following topics:

Feature description

With Bridging, the same DN can appear on up to eight single-line telephones. A maximum of five of these telephones can be equipped with ringers.

Incoming calls ring all telephones with a ringer connected and can be answered at any of the single-line telephones.

Operating parameters

A maximum of five C4A ringers are allowed on one parallel loop.

Feature interactions

Privacy

Privacy is lost when telephones are bridged. Any appearance of the DN can enter the call by going off-hook.

Feature packaging

This feature is included in base system software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 84: Busy Lamp Field

Contents

This section contains information on the following topics:

Feature description

When a DN is blocked due to the Attendant Blocking of Directory Number feature, the Busy Lamp Field/Enhanced Busy Lamp Field lamp corresponding to this DN displays the busy status of the DN as for ringing calls.

Busy Lamp Field/Console Graphics Module

The Busy Lamp Field/Console Graphics Module (BLF/CGM) is an add-on module for the Avaya 2250 Attendant Console. It can be configured to display the status of a specified 150 consecutive DNs (Standard Busy Lamp Field (SBLF), or all DNs, 100 at a time (Enhanced Busy Lamp Field [EBLF]). By monitoring the status, an attendant can tell a caller if the DN is busy prior to extending the call.

Enhanced Busy Lamp Field (EBLF) Array, displays the status of all DNs for a customer. The BLF/CGM displays the status of 100 DNs at a time on up to 63 Avaya 2250 Attendant Consoles. Each of the Console Graphics Modules can display a different hundreds group, while up to 20 CGMs can display the same hundreds group simultaneously.

When the attendant extends a call, a hundreds group is displayed after enough digits have been entered to determine the group. After a group has been established, the BLF/CGM shows the status for each DN in that group. Figure 18: Standard Busy Lamp Field on the BLF/CGM on page 589 shows an example of the EBLF on the BLF/CGM.

The EBLF continues to display the status of the hundreds group until another group is determined or until the module is cleared. The display is updated whenever the status of a DN in that group changes. The BLF is cleared when the attendant dials a new series of digits or releases the call.

Figure 18: Standard Busy Lamp Field on the BLF/CGM on page 589 shows the Standard Busy Lamp Field (SBLF) display on the CGM. The first and last DNs in the displayed group are listed as START EXT and END EXT. The START and END EXT DNs show the hundreds group displayed. The top row on the CGM designates the tens group. The left side shows the ones group. Figure 18: Standard Busy Lamp Field on the BLF/CGM on page 589 shows the busy DNs to be 3403, 3408, 3410, 3421, 3482, 3488, 3494, 3500, 3543, and 3549.

Figure 19: Enhanced Busy Lamp Field monitoring (example) on page 589 shows a system monitored by the EBLF. Each telephone represents a busy DN, listed beneath the telephone icon. The display screen at the top of the module defines the hundreds group as 35. The CGM displays the busy DNs within that group. The larger squares represent busy telephones within the group, and the smaller squares represent idle DNs. The attendant can quickly see which telephones are busy and which are idle.

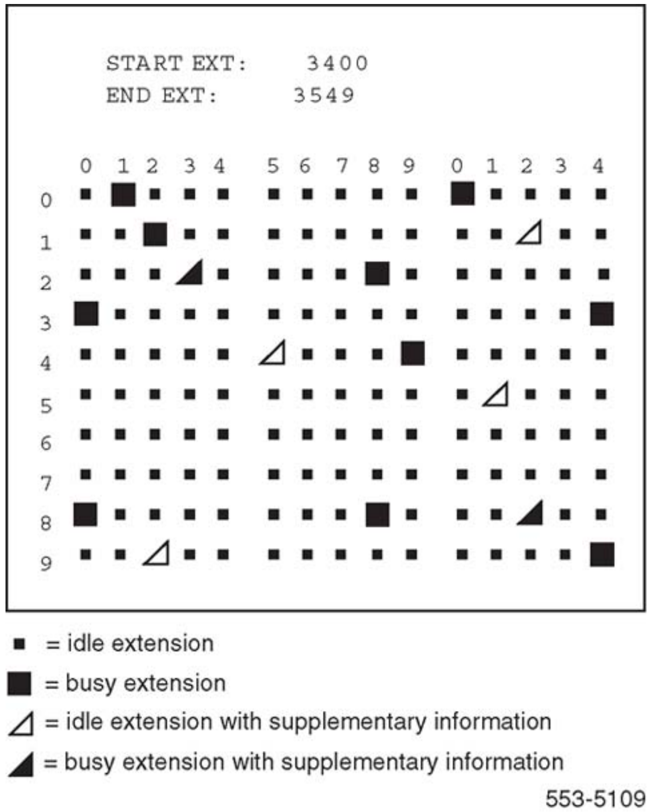


Figure 18: Standard Busy Lamp Field on the BLF/CGM

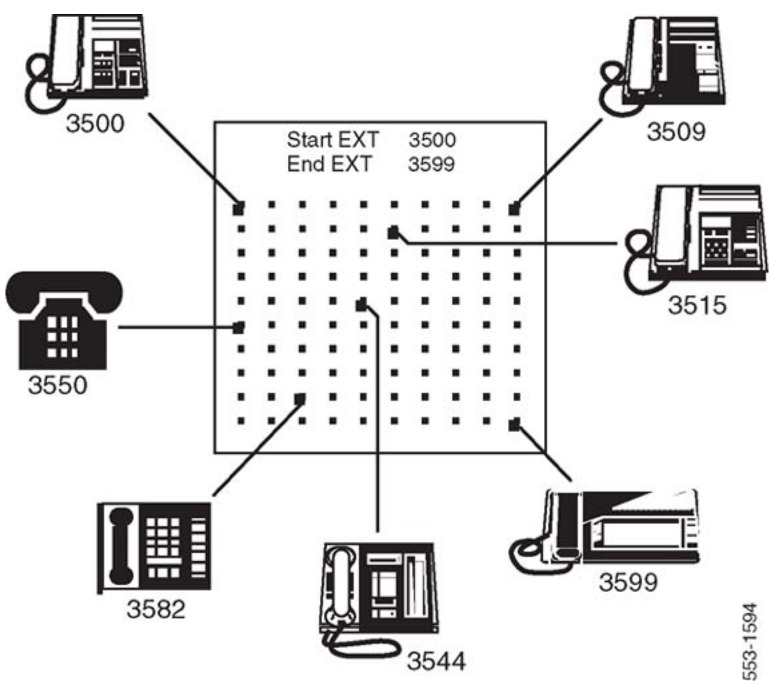


Figure 19: Enhanced Busy Lamp Field monitoring (example)

Operating parameters

Enough hundreds groups must be defined to support the maximum number of telephones to be monitored. The maximum number of hundreds is 99.

The EBLF requires an Avaya 2250 Attendant Console equipped with a BLF/CGM.

The SBLF and the EBLF are incompatible.

The EBLF supports mixed dialing plans (4, 5, 6, or 7 digits), but each hundreds group defined must be unique. For example, DNs 25XX and 25XXX cannot be configured in the same system. Any other DN group must begin with something other than 25 because, in this case, the CGM would be updated for DNs 2500 through 2599.

Only 20 attendant consoles can be updated for the same hundreds group simultaneously. If more than 20 consoles are monitoring the status of a single hundreds group, only the first 20 are updated. The remaining consoles display the earlier status, and an error message is output at this occurrence. (An unlimited number of consoles can be updated when they display different hundreds groups.)

When the Make Set Busy key is activated or deactivated, BLF updates only the first DN it finds on the attendant console. Lamp audit updates the status of subsequent DNs on the BLF.

Feature interactions

Attendant Blocking of Directory Number

When a DN is blocked due to the Attendant Blocking of DN feature, the Busy Lamp Field/ Enhanced Busy Lamp Field lamp corresponding to this DN displays the busy status of the DN as for ringing calls.

Call Park

A busy lamp field can be equipped to display the status of System Park DNs.

Idle Extension Notification

When an extension that is being supervised for an Idle Extension Notification to the attendant becomes idle, it is kept busy from receiving any incoming calls. The lamp on the attendant console for that DN will display a busy status, according to the parameters of the Busy Lamp Field/Enhanced Busy Lamp Field feature.

It is not possible to request Idle Extension Notification if the Busy Verify feature has been activated after the Busy Verify key is pressed.

Make Set Busy

When a Make Set Busy key is activated, the Busy Lamp Field array will indicate that the first DN only on that telephone is busy.

Feature packaging

- Busy Lamp Field Array (BLFA) is included in base system software.

Feature implementation

Task summary list

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

1. Table 197: LD 15 - Define the Busy Lamp Field/Console Graphics Module options in the Customer Data Block. on page 592
2. Table 198: LD 12 - Identify which attendant consoles have Enhanced Busy Lamp Field on the BLF/CGM. on page 592
3. Table 199: LD 10 - Activate DN hundreds groups for EBLF for each DN within each hundreds group. on page 592
4. Table 200: LD 11 - Activate DN hundreds groups for EBLF for each DN within each hundreds group. on page 593

Table 197: LD 15 - Define the Busy Lamp Field/Console Graphics Module options in the Customer Data Block.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	ATT_DATA	Attendant console options
CUST		Customer number
	0-99	Range for Large System and Avaya Communication Server 1000E system.
- OPT	(XBL) IBL	(Exclude) include Enhanced Busy Lamp Field.

Table 198: LD 12 - Identify which attendant consoles have Enhanced Busy Lamp Field on the BLF/CGM.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	2250	Attendant console type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
EBLF	(BLFD) BLFA	(Deny) allow Enhanced Busy Lamp Field.

When the BLF is configured before the telephones are programmed, the procedures in LD 10 and LD 11 are not required. As an alternative to reentering data when the BLF is configured after the telephones, a SYSLOAD associates the DN with the Hundreds Group (HGRP).

Table 199: LD 10 - Activate DN hundreds groups for EBLF for each DN within each hundreds group.

Prompt	Response	Description
REQ:	CHG	Change existing data.
TYPE:	500	Telephone type.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
DN	xxx...x	Reenter Directory Number (no change necessary).

Table 200: LD 11 - Activate DN hundreds groups for EBLF for each DN within each hundreds group.

Prompt	Response	Description
REQ:	CHG	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, Media Gateway 1000B, and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
KEY	xx aaa yyy...y	Reassign Directory Number (no change necessary), where: xx = key number aaa = DN type, and yyy...y = Directory Number.

Feature operation

To display the status of extensions on the BLF/CGM (attendant), follow these steps:

1. Press the **SHIFT** key, then the **conf. 6/BLF** key.

   The console is in the BLF mode.

2. Press the **Mode** key.

   The BLF/CGM screen displays the main menu.

3. Dial 0 (zero).

   The BLF/CGM displays the SBLF or the EBLF, depending on which option is configured in the system software.

Chapter 85: Busy Tone Detection for Asia Pacific and CALA

Contents

This section contains information on the following topics:

Feature description

The Busy Tone Detection feature for Asia Pacific and CALA uses the Digital Signaling Processor Universal Trunk (DXUT) card. This card is based on the Extended Universal Trunk card (EXUT) and allows for the following two capabilities:

- Flexible Busy Tone Detection
- Automatic Balance Impedance (AUTO_BIMP in Overlay 14)

The Flexible Busy Tone Detection functionality of this trunk card allows the system to recognize busy tones sent from a Public Exchange/Central Office. Busy Tone Detection permits disconnect supervision for Loop Start Central Office (CO) trunks. The Central Office provides busy tone to the last party involved in a call. The system detects this busy tone and disconnects the call.

Busy Tone Detection features are utilized in countries where tone detection is the only method for the system to detect far end disconnection.

The Busy Tone Detection feature for Asia Pacific and CALA uses the NT5D31 Digital Signaling Processor (DSP) Universal Trunk (DXUT) card. This card is based on the Extended Universal Trunk card (EXUT) and is configured in software as an EXUT card. However, the DXUT card has flexible busy tone detection provided by a Digital Signal Processor (DSP). The DXUT card also has tone detection intelligence that allows it to accurately differentiate between different disconnect tones sent by a Public Exchange/Central Office.

The DXUT card has programmable Busy Tone Detection characteristics which include:

- Cadence
- Incoming or Incoming and Outgoing call direction
- Tone Frequencies
- Tone Bandwidth
- Tone Levels

Tones are detected according to the parameters configured in Overlay 97.

When a trunk card does not support the Busy Tone Detection feature, it can still be configured in software; although, the hardware does not recognize the new Busy Tone messages. The DXUT messages are ignored by the old hardware.The existing hardware is still operational since the Busy Tone feature still supports the older hardware. Old messages are sent for backwards compatibility but are not resent to define frequency criteria.

The Automatic Balance Impedance (AUTO_BIMP) functionalities of the DXUT card enhance the Transhybrid Loss matching capability. The automatic balancing is performed by the Digital Signal Processor (DSP) when checking the reflections from the transmission line. When the software sends an AUTO_BIMP message to the DXUT card, the DSP generates a test tone and measures the amount of signal being reflected. The DSP then internally adjusts the balance network, in the codec, for the best Transhybrid loss.

Operating parameters

The Busy Tone Detection feature for Asia Pacific and CALA requires the DXUT card. The DXUT card requires busy tone detection data to be downloaded prior to activating this feature.

The AUTO_BIMP functionalities of this feature are not supported in the Digital Signaling Processor Universal Trunk (DXUT) card NT5D31 hardware.

Direct Inward Dialing (DID) trunks do not require busy tone supervision, since the Public Exchange/Central Office seizes the system trunk by closing the transmission loop. Far end trunk release is accomplished when the Public Exchange/Central Office opens the circuit.

Japan trunk cards, the Extended Universal Trunk card for Japan (XUTJ) and the Enhanced Extended Universal Trunk card for Japan (EXUTJ), do not support this feature. The DXUT card is not supported in Japan.

The system disconnects a call when a busy tone is detected on an incoming trunk. If the caller on the far end causes a busy tone to be generated, the call is disconnected, regardless of whether or not disconnection was intended. As an example, when a caller connected to a

Public Exchange/Central Office attempts to conference in a busy party, the system picks up this busy tone and the call is disconnected.

If any other types of tones (other than busy tone) are detected with the same cadence, frequency and level, the call is disconnected.

The Busy Tone Detection feature for Asia Pacific and CALA may not operate on conference bridges. In the scenario of Busy Tone Detection operating with a conference bridge, all of the trunks are incoming and an incoming Public Exchange/Central Office trunk disconnects from a conference. In this scenario, the disconnected trunk sends a busy tone signal to the conference bridge, and all trunks may be disconnected simultaneously.

In the event that an incoming call is connected to an external conference and two different Public Exchanges/Central Offices are sending busy tone signals at the same time, a stalemate condition may exist. When this occurs, the cadence of both busy tones may not be the same, and the resulted combination cadences may not be detected.

The DXUT card is based on the EXUT card design and is intended to operate in an EXUT-compatible Loss Planning environment. These EXUT compatible Loss Planning environments include the North American Loss Planning environment and Dynamic Loss Switching environments in certain countries.

Busy Tone characteristics are downloaded on a card basis. The Busy Tone Detection table assigned to the card is downloaded to the card when: the first trunk is configured, the card is disabled and enabled, the card is unplugged and reset, and during initialization after sysload, and when the Intelligent Peripheral Equipment package 203 is enabled.

Feature interactions

European XFCOT Support

When the XFCOT Busy Tone ID (BTID) is configured in Overlay 14 only the BTID is downloaded to the XFCOT card. The BTID is downloaded to the EXUT card when the Busy Tone Detection (BTD) package 294 is equipped.

Trunk to Trunk Connection

When the Trunk to Trunk Connection feature interacts with Busy Tone Detection for Asia Pacific and CALA, which ever feature occurs first takes precedence.

Timed Forced Disconnect

When Timed Forced Disconnect interacts with Busy Tone Detection for Asia Pacific and CALA, whichever feature occurs first takes precedence.

Feature packaging

Busy Tone Detection for Asia Pacific and CALA requires Busy Tone Detection (BTD) package 294.

Feature implementation

Task summary list

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

1. Table 201: LD 97 - Configure Busy Tone Detection (BTD) table parameters. on page 598
2. Table 202: LD 16 - Configure trunk units and trunk timers in the Route Data Block. on page 599
3. Table 203: LD 14 - Configure Busy Tone Supervision for a new Central Office Trunk. on page 600

Once the BTD table is configured, the new trunks can be entered and the required BTD table is assigned on a card basis. The BTD table number can only be entered in for the first unit programmed on the card.

Table 201: LD 97 - Configure Busy Tone Detection (BTD) table parameters.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	BTD	Busy Tone Detection.
BTDT	(0) - 7	Busy Tone Detection Table.
BCAD	(350) (350)	Busy Tone Cadence (in milliseconds). (ON cycle) (OFF cycle) (default)

Prompt	Response	Description
	500 500	For Japan. The values for each cycle are 0 to 1.5 seconds (1500 ms) and are entered in milliseconds. Input values are rounded to the nearest multiple of 25 ms. If zero (0) is entered for both phases, then a continuous tone occurs.
BTDD	(BOTH) INC	Busy Tone Detection Direction: Both Incoming and outgoing calls (default). Incoming calls only.
FREQ_0	350 - 655	Frequency of Busy Tone for Frequency 0 of a dual Busy Tone Detection to be detected in Hz. Valid entries are in multiples of 5Hz.
FREQ_1	350 - 655	Frequency of Busy Tone for Frequency 1 of a dual Busy Tone Detection to be detected in Hz. Valid entries are in multiples of 5Hz. For a single busy tone FREQ_1 must be set the same as FREQ_0.
FDLT	10 - 315	Frequency Delta. FDLT gives the tolerance of the tone to be detected in +/- hertz. Valid entries are in multiples of 5Hz. For dual Busy Tone Detection on the NT5D31 card, the same maximum and minimum levels apply to both tones.
FLVL_MAX	0 - 15	Maximum Frequency Tone level to be detected. Valid entries are in multiples of 5dBm. For dual Busy Tone Detection on the NT5D31 card, the same level applies to both tones.
FLVL_MIN	20 - 35	Minimum Frequency Tone level to be detected. Valid entries are in multiples of 5dBm. For dual Busy Tone Detection on the NT5D31 card, the same level applies to both tones.

Table 202: LD 16 - Configure trunk units and trunk timers in the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add a new data block to the system.
TYPE	RDB	Define a new Route Data Block.
CUST	xx	Customer number, as defined in LD 15
ROUT		Route number
	0-511	Range for Large System and Avaya Communication Server 1000E system.
TKTP	COT	Define trunk type as Central Office.
ICOG	IAO	Incoming and Outgoing trunk.
CNTL	YES	Changes to controls or timers.

Prompt	Response	Description
NEDC	ETH	Either end control.
FEDC	ETH	Either end control.

Table 203: LD 14 - Configure Busy Tone Supervision for a new Central Office Trunk.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	COT	Central Office trunk.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	EXUT	Type is IPE EXUT. This includes the DXUT. (This prompt is required only for the first unit defined on each card.)
CUST	xx	Customer number, as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
SIGL	LOP	Loop start level 3 signaling.
TIMP	(600) 900	Termination Impedance.
BIMP	(3COM) 3CM2 600 900	Balance Impedance. In the case of AUTO_BIMP, this BIMP value is used as a default value if an optimum AUTO_BIMP is not found or if the AUTO_BIMP test is not complete.
AUTO_BIMP	YES	Automatic Balance Impedance is set according to transmission line parameters. NO = default for new trunks.
SUPN	YES	Answer and disconnect supervision required.
-STYP	PIP BTS PIP BTS	Supervision Type. Polarity Insensitive Pack. Busy Tone Supervision. Both options.
BTDT	(0)-7	Busy Tone Detection Table number configured in LD 97. (This prompt is required only for the first unit defined on each card.)
CLS	(DIP) DTN	Dial Pulse. Digitone.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 86: Busy Tone Detection for Japan

Contents

This section contains information on the following topics:

Feature description

In many countries, Central Office loop start trunks are not supervised. This can lead to difficulties for incoming calls to the system that require disconnect supervision to operate properly. Through a modification to the tone detector, this feature allows the system to perform disconnect supervision through the recognition of a busy tone sent by the Public Exchange/ Central Office.

Busy Tone Detection for Japan allows a technician to enter the characteristics of the busy tone tables in LD 97. When these characteristics are programmed, the information is downloaded to the system during call processing. When a busy tone is detected, the trunk sends a message to the system software to disconnect the call and free the trunk for other uses.

This feature provides Japan Central Office (JCO) and Japan Direct Inward Dialing (JDID) trunks with Busy Tone Detection (BTD) capability through trunk supervision.

Operating parameters

The feature is applicable to all systems.

Busy Tone Detection for Japan requires the Enhanced Extended Universal Trunk Card for Japan (EXUTJ).

This feature requires a busy tone from the Public Exchange/Central Office.

The system disconnects any call if a busy tone is detected on the incoming trunk. If called party causes a busy tone to be generated, the call disconnects whether intended or not. As an example, this may happen if a Central Office user tries to conference in a busy party. The busy tone is detected by the circuit switched network trunk and the call disconnects.

If another tone is configured similar to the Busy Tone (frequency + or - 30 Hz and cadence within + or - 100 ms), the busy tone detector is interpreted as a busy tone and the call is disconnected. Therefore, tones should be configured so they can be interpreted correctly.

The busy tone detection characteristics are downloaded on a card basis only. All units on the trunk card must go to the Central Office that produces the same Busy Tone cadence.

To modify the busy tone detection table assigned to a trunk card, all trunks on that card must be removed initially from the software (LD 14). It is recommended that all Central Office loop start trunk units be on the same card and configured in the same route.

500/2500 Line Disconnect Supervision is supported by this feature.

If the trunk card is not designed to support the Busy Tone Detection (BTD) feature, BTD can still be configured in the software. However, no feedback is given to the technician that a discrepancy exists between the software and hardware configuration.

Feature interactions

Timed Forced Disconnect

Busy Tone Detection for Japan activates a timer to start once a Central Office (as well as other types of trunks) has been seized. After this timer expires, the trunk is forced to disconnect. BTD does not impact this timer; however, whichever timer occurs first will prevail.

Trunk to Trunk Connection

Busy Tone Detection for Japan does not impact the Trunk to Trunk Connection feature. However, which ever occurs first prevails.

Feature packaging

Busy Tone Detection for Japan is Busy Tone Detection (BTD) package 294.

The following packages are also required:

- Japan Central Office Trunk (JPN) package 97
- Intelligent Peripheral Equipment (XPE) package 203

Feature implementation

Task summary list

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

1. Table 204: LD 97 - Assign Tone Characteristics to Busy Tone Detection Tables. on page 603
2. Table 205: LD 14 - Assign Busy Tone Detection to Central Office (CO), Foreign Exchange (FEX) and WATS Trunks. on page 604
3. Table 206: LD 14 - Assign Busy Tone Detection to Direct Inward Dialing (DID) Trunks. on page 604

Table 204: LD 97 - Assign Tone Characteristics to Busy Tone Detection Tables.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	BTD	Busy Tone Detection data block.
BTDT	(0)-7 X1-X7	Busy Tone Detection table. Table 0 can be changed but cannot be removed. Table 0 should always exist (when the BTD package is equipped) and is initialized to the default value for Japan. When creating alternate tables, table 0's values are used to fill the table and these can be changed. Enter X in front of the table number to remove the table.
BCAD	500 500 (ph1 ph2)	Busy Tone Cadence (on and off phase length during the cycle can be entered). ph1 is the ON cycle and ph2 is the OFF cycle. The values for each phase can be 0 to 1.5 seconds (1500 ms) and are entered as ms. The input values are rounded to the nearest multiple of 25 ms.

Prompt	Response	Description
		Entering all 0s indicates continuous tone. A tone is deemed continuous if it lasts for at least 3.2 seconds. The smallest cadence is 50 ms even though 25 ms can be entered.
BTDD	(BOTH)	Busy Tone Detection Direction. BOTH = both incoming and outgoing calls
	INC	INC = incoming calls only

Table 205: LD 14 - Assign Busy Tone Detection to Central Office (CO), Foreign Exchange (FEX) and WATS Trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	COT	Central Office Trunk.
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya Communication Server 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XUT	Enhanced Extended Universal Trunk.
...		
SIGL	LOP	Loop start signaling.
...		
SUPN	YES	Answer and disconnect supervision required. If SUPN = YES, then the values stored in supervision type (STYP prompt) are initialized and only the current entered values are saved. Therefore, complete supervision is required every time through this branch.
- STYP	xxx	Trunk supervision type where xxx is: PIP = Polarity Insensitive JCO = Japan Central Office BTS = Busy Tone Supervision
...		
BTDT	(0)-7	Busy Tone Detection Table. This table must be defined in LD 97.

Table 206: LD 14 - Assign Busy Tone Detection to Direct Inward Dialing (DID) Trunks.

Prompt	Response	Description
REQ	NEW CHG	Add new data. Change existing data.
TYPE	DID	Direct Inward Dialing Trunk.

Prompt	Response	Description
TN	l s c u	Terminal Number Format for Large System, Media Gateway 1000B, and Avaya CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
XTRK	XUT	Enhanced Extended Universal Trunk.
...		
SIGL	LOP	Loop start signaling.
...		
SUPN	YES	Supervision. This response is automatically prompted YES for DID LOP.
- STYP	xxxx	Trunk supervision type where xxxx is: JDID = Japan DID. When XTRK = XUT and <CR> is entered STYP default to JDID. JDID BTS = Busy Tone Supervision and JDID (XUT only). When XTRK = XUT and BTS is entered STYP defaults to JDID BTS.
...		
BTDT	(0)-7	Busy Tone Detection Table. This table must be defined in LD 97.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 87: Busy Verify on Calling Party Control Calls

Contents

This section contains information on the following topics:

Feature description

This enhancement to the Busy Verify feature changes the way in which a local attendant and toll attendant, and Network Attendant Service attendant are able to Busy Verify, Barge-In, and Break-In to a station that is connected to a trunk on a route that has Calling Party Control (CGPC) set to YES.

Table 207: Busy verify on calling party control calls operation for a local call2

	Busy Verify	Barge-In	Break-In
Local attendant	Yes	Yes	Yes
Toll attendant	–	–	Yes
NAS attendant	–	–	Yes

Table 208: Title: Busy verify on calling party control calls operation for a toll call

	Busy Verify	Barge-In	Break-In
Local attendant	No	No	No
Toll attendant	–	–	No

	Busy Verify	Barge-In	Break-In
NAS attendant	–	–	No

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

Attendant Break-In

Local Attendant Break-In will be temporarily denied if the desired party is already in a toll operator Break-In conference or on a Special Service call, or awaiting the Special Operator signal. Local attendant/toll operator Break-In will be temporarily denied if the desired party is established on an incoming toll call.

Network Attendant Services (NAS)

A NAS attendant is not allowed to Busy Verify to a station on a different node, or Barge-In to a trunk on a different node. A NAS attendant is allowed to Break-In to a station on a different node, if the incoming trunk on the route is not a toll call. NAS attendant Break-In will be temporarily denied if the desired party is already on a toll call, a toll operator Break-In conference, or a Special Service call, or awaiting the Special Operator signal.

Feature packaging

Busy Verify on Calling Party Control Calls requires Operator Call Back (OPCB) package 126.

Feature implementation

No change to existing configuration is required for the Busy Verify on Calling Party Control Calls feature.

Feature operation

See the following feature descriptions contained within this document.

- Attendant Busy Verify on page 271
- Attendant Barge-In on page 218
- Attendant Break-In on page 235